(19) Europäisches Patentamt
European Patent Office
Office européen des brevets

(11) EP 0 637 803 B1

(12) EUROPEAN PATENT SPECIFICATION

(45) Date of publication and mention of the grant of the patent:
06.12.2000 Bulletin 2000/49

(51) Int. Cl.[7]: **G06F 17/17**, G06F 19/00

(21) Application number: **94111471.2**

(22) Date of filing: **22.07.1994**

(54) Method and device for adaptively estimating a transfer function of an unknown system

Verfahren und Vorrichtung zur adaptiven Schätzung einer Übertragungsfunktion eines unbekannten Systems

Méthode et dispositif d'estimation adaptatif d'une fonction de transfert d'un système inconnu

(84) Designated Contracting States:
DE FR

(30) Priority: **27.07.1993 JP 18474293**

(43) Date of publication of application:
08.02.1995 Bulletin 1995/06

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Description

BACKGROUND OF THE INVENTION

[0001] The present invention relates to a method for adaptively estimating, with a projection algorithm, a transfer function of an unknown system and its output in an acoustic canceller, active noise control or the like and an estimating device using such a method.

[0002] In the following description, time will be represented by a discrete time k. For example, the amplitude of a signal x at time k will be expressed by x(k). Fig. 1 is a diagram for explaining the estimation of the transfer function of an unknown system. Reference numeral 11 denotes a transfer function estimation part and 12 the unknown system, and reference character x(k) represents an input signal to the unknown system and y(k) an output signal therefrom. The transfer function h(k) of the unknown system is estimated using the input signal x(k) and the output signal y(k). Fig. 2 is a diagram for explaining an adaptive estimation of the transfer function. Reference numeral 21 denotes an estimated transfer function correcting vector calculation part, 22 an estimated transfer function correction part and 23 a convolution part, these parts constituting an estimated signal generation part 20. The transfer function h(k) of the unknown system 12 is estimated as a transfer function $\hat{h}(k)$ of an FIR filter of a tap number L which forms the convolution part 23. More specifically, coefficients $\hat{h}_1(k), ..., \hat{h}_L(k)$ of the FIR filter are estimated. Let it be assumed that the "transfer function" and the "FIR filter coefficient" will hereinafter be construed as the same. For the sake of brevity, the filter coefficient is represented as an estimated transfer function vector $\hat{h}(k)$ defined by the following equation.

$$\hat{h}(k) = [\hat{h}_1(k), \hat{h}_2(k), ..., \hat{h}_L(k)]^T \quad (1)$$

where T represents a transpose.

[0003] In Fig. 2, the input signal x(k) to the unknown system 12 is fed to the convolution part 23 and a calculation is performed to obtain the estimated transfer function vector $\hat{h}(k)$ that minimizes an expected value which is the square of an error signal e(k) available from a subtractor 24 which detects a difference between an output $\hat{y}(k)$ from the convolution part 23 given by the following equation (2) and the output y(k) from the unknown system 12.

$$\hat{y}(k) = \hat{h}(k)^T x(k) \quad (2)$$

$$x(k) = [x(k), x(k-1), ..., x(k-L+1)]^T \quad (3)$$

where $\hat{y}(k)$ is an estimated value of the output from the unknown system, which is close to the value of the output y(k) when the estimated transfer function $\hat{h}(k)$ is close to an unknown characteristic.

[0004] In practice, the transfer function of an unknown system often varies with time as in the case where the transfer function of an acoustic path varies with movement of audiences or objects in a sound field such as a conference hall or theater. For this reason, the estimated transfer function $\hat{h}(k)$ of the unknown system also needs to be adaptively corrected in accordance with a change in the acoustic path of the unknown system. The estimated transfer function correcting vector calculation part 21 calculates an estimated transfer function correcting vector $\delta\hat{h}(k)$ on the basis of the error signal e(k) and the input signal x(k) to the unknown system 12. The estimated transfer function correction part 22 corrects the estimated transfer function by adding the estimated transfer function correcting vector $\delta\hat{h}(k)$ to the estimated transfer function vector $\hat{h}(k)$ as expressed by the following equation.

$$\hat{h}(k+1) = \hat{h}(k) + \mu\delta\hat{h}(k) \quad (4)$$

where μ is called a step size, which is a preselected quantity for controlling the range of each correction and is handled as a time-invariant constant. In the following description, the estimated transfer function correcting vector $\delta\hat{h}(k)$ is calculated for $\mu = 1$ and, if necessary, it is multiplied by a desired step size μ. In some applications the value of the step size μ is caused to vary with time, but in such a case, too, the following description is applicable. The corrected estimated transfer function vector is transferred to the convolution part 23. The above is a transfer function estimating operation at time k and the same operation is repeated after time k + 1 as well.

[0005] T. Furukawa et al: "The Orthogonal Projection Algorithm for Block Adaptive Signal Processing" in Proceedings of ICASSP 98, IEEE New York USA, Vol. 2, 23-26 May 1989, Glasgow UK, pp. 1059-1062, disclose estimation of a transfer function of an unknown system by using orthogonal projection based on an Affine Projection Algorithm. The aim is to reduce the number of multiplications, wherein a filter coefficient vector is updated for each block of input vectors (or every r sample times) to reduce computational requirement. Since the coefficient vector is updated for each block L, the number of update times is reduced, resulting in decrease in convergence speed of the adaptive filter. The filter coefficient vector is adjusted according to an equation numbered 11 in the document and including the Moore-Pen-

rose type generalized inverse matrix which still involves a high computational requirement. The projection algorithm proposed by T. Furukawa et al. is developed from the Affine Projection Algorithm proposed by Ozeki and Umeda and mentioned below.

[0006] The estimated transfer function correcting method described above in respect of Fig. 2 is known as an adaptive algorithm. While an LMS (Least Mean Square) algorithm and an NLMS (Normalized LMS) are also well-known as adaptive algorithms, a description will be given of a projection algorithm proposed in a literature [Ozeki and Umeda: "An Adaptive Filtering Algorithm Using an Orthogonal Projection to an Affine Subspace and Its Properties", Journal of Institute of Electronics, Information and Communication Engineers of Japan (A), J67-App. 126-132, (1984-2).] This document discloses a method of adaptively estimating thetransfer function of an unknown system in accordance with the first portion of claims 1, 2 and 3.

[0007] The projection algorithm requires a larger number of operations than does the NLMS but has an excellent adaptability to a speech signal input. With the projection algorithm, as referred to above, the vector $\delta\hat{h}(k)$ is determined by Eq. (4) for $\mu = 1$ so that simultaneous equations composed of the following p equations are satisfied.

$$\begin{aligned} y(k) &= x(k)^T(\hat{h}(k) + \delta\hat{h}(k)) \\ y(k-1) &= x(k-1)^T(\hat{h}(k) + \delta\hat{h}(k)) \\ &\vdots \\ y(k-p+1) &= x(k-p+1)^T(\hat{h}(k) + \delta\hat{h}(k)) \end{aligned} \qquad (5)$$

Eq. (5) indicates that the vector $\delta\hat{h}(k)$ is determined so that the estimated transfer function $\hat{h}(k+1)$ updated at time k provides the same values y(k), y(k-1), ..., y(k-p+1) as the outputs from the unknown system, respectively, for p input vectors x(k), x(k-1), ..., x(k-p+1) prior to time k. By this, it is expected that the characteristic of the estimated transfer function $\hat{h}(k)$ will approach the characteristic of the unknown system as the adaptive updating of the estimated transfer function is repeated using the vector $\delta\hat{h}(k)$.

[0008] In the above, p is a quantity commonly called a projection order. As the projection order p increases, the adaptability of the projection algorithm increases but the computational complexity also increases. The conventional NLMS method corresponds to the case of p = 1.

[0009] Now, transposing the first equation in Eq. (5), we have

$$x(k)^T\delta\hat{h}(k) = y(k) - x(k)^T\hat{h}(k) = e(k) \qquad (6)$$

Furthermore, transposing the second equation in Eq. (5) and using an equation obtainable by setting k in Eqs. (4) and (6) to k-1, we have

$$\begin{aligned} x(k-1)^T\delta\hat{h}(k) &= y(k-1) - x(k-1)^T\hat{h}(k) \\ &= y(k-1) - x(k-1)^T\hat{h}(k-1) + \mu\delta\hat{h}(k-1)) \\ &= y(k-1) - x(k-1)^T\hat{h}(k-1) + \mu x(k-1)^T\delta\hat{h}(k-1) \\ &= e(k-1) - \mu e(k-1) \\ &= (1-\mu)e(k-1) \end{aligned} \qquad (7)$$

Thereafter, the following relationship similarly holds.

$$x(k-i)^T\delta\hat{h}(k) = (1-\mu)^i e(k-i) \qquad (8)$$

Based on this, Eq. (5) can be rewritten as the following system of simultaneous equations.

$$X_p(k)\delta\hat{h}(k) = e(k) \qquad (9)$$

where $X_p(k)$ is a matrix with p rows and L columns and e(k) is a vector of the p rows; they are defined by the following equations.

$$\mathbf{x}_p(k) = \begin{bmatrix} \mathbf{x}(k)^T \\ \mathbf{x}(k-1)^T \\ \mathbf{x}(k-p+1)^T \end{bmatrix} \quad (10)$$

$$\mathbf{e}(k) = \begin{bmatrix} e(k) \\ (1-\mu)e(k-1) \\ \vdots \\ (1-\mu)^{p-1}e(k-p+1) \end{bmatrix} \quad (11)$$

The vector e(k) will hereinafter be referred to as an error vector. Now, since p is usually smaller than L, Eq. (9) is an under-determined simultaneous equation or indeterminate equation for the vector $\delta\hat{\mathbf{h}}(k)$ and the minimum norm solution of the vector $\delta\hat{\mathbf{h}}(k)$ is given by the following equation.

$$\begin{aligned} \delta\hat{\mathbf{h}}(k) &= \mathbf{X}_p(k)^T(\mathbf{X}_p(k)\mathbf{X}_p(k)^T)^{-1}\mathbf{e}(k) \\ &= [x(k)x(k-1), \ldots, x(k-p+1)]\mathbf{g}(k) \end{aligned} \quad (12)$$

where

$$\mathbf{g}(k) = \mathbf{R}_p(k)^{-1}\mathbf{e}(k) \quad (13)$$

$$\mathbf{R}_p(k) = \mathbf{X}_p(k)\mathbf{X}_p(k)^T \quad (14)$$

$\mathbf{R}_p$ is a matrix with p rows and p columns, which will hereinafter be referred to as a p-order covariance matrix or auto-correlation matrix, and g(k) a pre-filter coefficient vector. Letting elements of the pre-filter coefficient vector g(k) be represented by $g_1(k)$, $g_2(k)$, ..., $g_p(k)$, the estimated transfer function correcting vector $\delta\hat{\mathbf{h}}(k)$ can be expressed on the basis of Eq. (12) as follows:

$$\delta\hat{\mathbf{h}}(k) = \sum_{i=1}^{p} g_i(k)\, x(k-i+1) \quad (15)$$

Thus, when the projection algorithm is used, in accordance with the present invention the estimated transfer function correcting vector calculation part 21 in Fig. 2 has such a construction as depicted in Fig. 3. Reference numeral 31 denotes a pre-filter coefficient vector calculation part, which uses the input signal x(k) and the error signal e(k) to calculate the pre-filter coefficient g(k) on the basis of Eq. (13). Reference numeral 32 denotes a pre-filtering part, which performs the pre-filtering operation expressed by Eq. (15) to synthesize the estimated transfer correcting vector $\delta\hat{\mathbf{h}}(k)$ by use of the pre-filter coefficient g(k) that is transferred from the pre-filter coefficient vector calculation part 31.

[0010] Now, a description will be given of the computational complexity of the projection scheme described above. The computational complexity mentioned herein is the number of multiplication-addition (or addition) operations necessary for estimating operations per unit discrete time. The computational complexity of Eq. (2) in the convolution part 23 of the tap number L in Fig. 2 is L. The computational complexity of Eq. (13) in the pre-filter coefficient vector calculation part 31 of the estimated transfer function correcting vector calculation part 21 is about $p^3/6$ when using the Choleski method which is a typical computation method. The computational complexity of Eq. (15) in the pre-filtering part 32 is (p-1)L. The computational complexity of Eq. (4) in the estimated transfer function correction part 22 is L. Thus, the entire computational complexity NC of the projection scheme is given by the following equation.

$$NC = L + p^3/6 + (p-1)L + L \quad (16)$$

[0011] On the other hand, the computational complexity of the NLMS scheme or LMS algorithm is about 2L. For example, when L = 500 and p = 20 (a typical value in the case of an acoustic echo canceler), the number of operations involved in the NLMS scheme is 1000, whereas the projection scheme requires as many as about 12000 operations on the basis of Eq. (16). The computational complexity $p^3/6$ of Eq. (13), in particular, abruptly increases as the projection order p increases. Thus, the projection scheme has excellent convergence characteristics as compared with the NLMS scheme but poses the problem of increased computational complexity.

SUMMARY OF THE INVENTION

[0012] It is therefore an object of the present invention to provide an adaptive transfer function estimating method which permits reduction of the computational complexity involved and an estimating device using such a method.

[0013] This object is achieved with a method as claimed in claim 1 or 2 and a device as claimed in claim 7 or 8. Preferred embodiments are subject-matter of the depnedent claims.

[0014] To attain the above object, a first embodiment of the present invention reduces the computational complexity of the pre-filter coefficient vector calculation part 31 while a second embodiment and/or reduces the computational complexity of the pre-filtering part 32.

BRIEF DESCRIPTION OF THE DRAWINGS

[0015]

Fig. 1 is a block diagram illustrating an ordinary construction for the estimation of a transfer function;
Fig. 2 is a block diagram of a transfer function estimation part 11 in Fig. 1;
Fig. 3 is a block diagram of an estimated transfer function correcting vector calculation part 21 embodying the present invention in Fig. 2;
Fig. 4 is a block diagram of a pre-filter coefficient vector calculation part 31 according to the first aspect of the invention in Fig. 3;
Fig. 5 is a block diagram showing an estimated signal generating process according to the second aspect of the invention;
Fig. 6 is a block diagram showing an application of the present invention to measurement of the transfer function of a loudspeaker;
Fig. 7 is a block diagram showing an application of the present invention to an echo canceller;
Fig. 8 is a block diagram showing an application of the present invention to noise control;
Fig. 9 is a graph showing echo cancelling characteristics of an echo canceller embodying the present invention; and
Fig. 10 is a graph showing, in comparison with a prior art example, the relationship between the number of operations for the estimation of the transfer function and the projection order.

DESCRIPTION OF THE PREFERRED EMBODIMENTS

[0016] A description will be given first of a method for reducing the computational complexity of the pre-filter coefficient vector calculation part 31. This method is characterized in that the pre-filter coefficient vector g(k) is obtained by a recursion formula utilizing a vector g(k-1) at one unit time before. Now, it will be demonstrated how the vector g(k) can be obtained from the immediately preceding vector g(k-1) on a recursive basis. The recursion formula is derived through utilization of the fact that elements of the error vector e(k) of Eq. (11) shift with time while being multiplied by 1-μ and the property of the inverse of the covariance matrix of the input signal given by Eq. (14).

[0017] As is the case with Eq. (14), a covariance matrix $R_{p-1}(k)$ of the p-1 order is defined by the following equation.

$$R_{p-1}(k) = x_{p-1}(k)x_{p-1}(k)^T$$

$$= \begin{bmatrix} x(k)^T \\ x(k-1)^T \\ \vdots \\ x(k-p+2)^T \end{bmatrix} [x(k), x(k-1), \ldots, x(k-p+2)] \qquad (17)$$

[0018] It is known from a literature, S. Haykin, "Adaptive Filter Theory," 2nd edition, Prentice-Hall, 1991, pp. 577-578 that the inverse of the covariance matrix $R_p(k)$ and the inverse of the covariance matrix $R_{p-1}(k)$ bear such a relationship as shown below.

$$R_p(k)^{-1} = \begin{bmatrix} 0 & \cdots & 0 \\ \vdots & R_{p-1}(k-1)^{-1} & \\ 0 & & \end{bmatrix} + \frac{a(k)a(k)^T}{F(k)} \qquad (18)$$

$$R_p(k-1)^{-1} = \begin{bmatrix} & & 0 \\ & R_{p-1}(k-1)^{-1} & \vdots \\ & & 0 \\ 0 & \cdots & 0 \end{bmatrix} + \frac{b(k-1)b(k-1)^T}{B(k-1)} \qquad (19)$$

where a(k) is a forward linear prediction coefficient vector (p-order) which satisfies a normal equation $R_p(k)a(k) = [F(k), 0, ..., 0]^T$ and its first element is a 1; F(k) is the minimum value of the sum of forward a posteriori prediction-error squares; b(k-1) is a backward linear prediction coefficient vector (p-order) which satisfies a normal equation $R_p(k-1)b(k-1) = [0, ..., 0, B(k-1)]^T$ and its last element is a 1; and B(k-1) is the minimum value of the sum of backward a posteriori prediction-error squares. The linear prediction coefficients a(k), b(k-1) and the minimum values of the sums of forward and backward a posteriori prediction-error squares F(k) and B(k-1) could be calculated with less computational complexity through use of an FTF (Fast Transversal Filters) algorithm, for instance, as disclosed in J. M. Cioffi and T. Kailath, "Windowed fast transversal adaptive filter algorithms with normalization," IEEE Trans. Acoust, Speech Signal Processing, vol. ASSP-33, no. 3, pp. 607-625. The pre-filter coefficient vector g(k) of Eq. (13) is shown again below.

$$g(k) = R_p(k)^{-1}e(k) \qquad (20)$$

A pre-filter deriving coefficient vector f(k) of p-1 order is defined, corresponding to the vector g(k), by the following equation.

$$f(k) = R_{p-1}(k)^{-1}e_{p-1}(k) \qquad (21)$$

where

$$\bullet_{p-1}(k) = \begin{bmatrix} e(k) \\ (1-\mu)e(k-1) \\ \vdots \\ (1-\mu)^{p-2}e(k-p+2) \end{bmatrix} \qquad (22)$$

Since all elements of a vector $(1-\mu)e_{p-1}(k-1)$ of p-1 order, obtained by substituting (k-1) for k in Eq. (22) and multiplying both of its left and right sides by $(1-\mu)$, constitute elements of the p-order vector e(k) of Eq. (11) except its first element e(k), the relationship of the following equation holds.

$$\bullet(k) = \begin{bmatrix} e(k) \\ (1-\mu)\bullet_{p-1}(k-1) \end{bmatrix} \qquad (23)$$

Furthermore, since all the elements of the p-1 order vector of Eq. (22) constitute all elements of the p-order vector of Eq. (11) except its last element, the relationship between the vectors e(k) and $e_{p-1}(k)$ can also be expressed as follows:

$$e(k) = \begin{bmatrix} \bullet_{p-1}(k) \\ (1-\mu)^{p-1}e(k-p+1) \end{bmatrix} \qquad (24)$$

Multiplying the respective terms on both sides of Eq. (18) by e(k) from the right, we obtain the following equation from Eqs. (20), (21) and (23).

$$g(k) = (1-\mu)\begin{bmatrix} 0 \\ f(k-1) \end{bmatrix} + \frac{a(k)^T e(k)}{F(k)} a(k) \qquad (25)$$

Moreover, multiplying both sides of Eq. (19) by e(k-1) from the right, we obtain the following equation from Eqs. (20), (21) and (24).

$$g(k-1) = \begin{bmatrix} f(k-1) \\ 0 \end{bmatrix} + \frac{b(k-1)^T e(k-1)}{B(k-1)} b(k-1) \qquad (26)$$

Transposing the right side to the left side, we have

$$\begin{bmatrix} \mathbf{f}(k-1) \\ 0 \end{bmatrix} = \mathbf{g}(k-1) - \frac{\mathbf{b}(k-1)^T \mathbf{e}(k-1)}{B(k-1)} \mathbf{b}(k-1) \qquad (27)$$

In this way, the pre-filter coefficient vector g(k) is calculated on the basis of the value of the pre-filter deriving coefficient vector f(k-1) as expressed by Eq. (25), and the vector f(k-1) is calculated from the vector g(k-1) as expressed by Eq. (27). That is, Eqs. (25) and (27) are recursion formulae for the vector g(k).

[0019] On the right side of Eq. (25), to obtain the first term (1-μ)[ ] needs p-1 multiplications to obtain e(k) needs p-1 multiplications and to obtain the second term $a(k)^T e(k)$ needs p multiply-add operations, one division for division by F(k) and p multiply-add operations for multiplication by a(k) and addition to the first term on the right side. That is, a total of 4p operations are needed. Similarly, approximately 2p operations are required to obtain Eq. (27). Accordingly, the number of operations of Eqs. (25) and (27) is around 6p, which is a computational complexity proportional to p. On the other hand, it is disclosed in the afore-mentioned literature by J. M. Cioffi and T. Kailath that the linear prediction coefficient vectors a(k), b(k-1) and the minimum values of the sum of a posteriori prediction-error squares F(k), B(k-1), which are needed in Eqs. (25) and (27), can be calculated by a linear prediction scheme with a computational complexity of about 10p. Thus, it is evident that the recursion formulae of Eqs. (25) and (27) permits calculation of the vector g(k) with the computational complexity proportional to the projection order p.

[0020] According to the present invention based on the above-described principles, the matters (A), (B) and (C) listed below are effective in reducing the computational complexity and in increasing the computational stability.

(A) Eq. (27) is an equation for the p-order vector and its left side is zero with respect to a p-th element (the least significant element) of the vector. Since a p-th element of the vector b(k-1) is always a 1, the following equation holds using a p-th element $g_p(k-1)$ of the vector g(k-1).

$$\frac{\mathbf{b}(k-1)^T \mathbf{e}(k-1)}{B(k-1)} = g_p(k-1) \qquad (28)$$

Therefore, the following equation may be calculated in place of Eq. (27).

$$\begin{bmatrix} \mathbf{f}(k-1) \\ 0 \end{bmatrix} = \mathbf{g}(k-1) - g_p(k-1)\, \mathbf{b}(k-1) \qquad (27a)$$

In this case, however, taking computational errors into account, it may sometimes be advantageous to obtain both $g_p(k-1)$ and the left side of Eq. (28) and average them.

(B) When denominators F(k) and B(k-1) of the second terms on the right sides of Eqs. (25) and (27) are small, operations are unstable. The computational instability could be reduced by adding non-negative $\delta_F(k)$ and $\delta_B(k-1)$ to the denominators as shown in the following equations.

$$g(k) = (1-\mu)\begin{bmatrix} 0 \\ f(k-1) \end{bmatrix} + \frac{a(k)^T e(k)}{F(k) + \delta_F(k)} a(k) \qquad (29)$$

$$\begin{bmatrix} f(k-1) \\ 0 \end{bmatrix} = g(k-1) - \frac{b(k-1)^T e(k-1)}{B(k-1) + \delta_B(k-1)} b(k-1) \qquad (30)$$

In practice, the values $\delta_F(k)$ and $\delta_B(k-1)$ may be set to desired values about 40 dB smaller than the average power of the input signal x(k) (that is, about 1/10000 the average power) or may also be varied with time k in accordance with power variations.

(C) In linear prediction analysis for obtaining the linear prediction coefficient vectors **a**(k), **b**(k-1) and the minimum values of the sum of prediction-error squares F(k), B(k-1) which are needed in Eqs. (25) and (27), the analysis frame of the input signal x(k) differs from that in an ordinary case. In concrete terms, the analysis frame in an ordinary linear prediction analysis ranges from time 0 to the current time, and when time k-1 is updated to k, x(k) needs only to be added to the analysis frame. In the projection algorithm, the analysis frame ranges from x(k) to x(k-L-p+2) in Eq. (5); so that when time k-1 is updated to k, it is necessary not only to add x(k) to the analysis frame but also to remove x(k-L-p+1). On this account, the linear prediction analysis in the projection algorithm requires computational complexity twice that needed for ordinary linear prediction analysis. However, when the statistical property of the input signal does not change or when it is expected that its change is slow, the result of the linear prediction analysis does not largely depend on the analysis frame; hence, it is possible to use the ordinary linear prediction analysis in which x(k) needs only to be added to the analysis frame at time k -- this permits reduction of the computational complexity.

[0021] Next, a description will be given of an embodiment of the adaptive transfer function estimating method of the present invention based on the above-described theoretical discussions, reference being made to Fig. 2 because the overall construction of its functional block is the same as shown in Fig. 2. Since the above-described theory underlying the present invention is related to the reduction of computational complexity in the estimated transfer function correcting vector calculation part 21 in Fig. 2, in particular, in the pre-filter coefficient vector calculation part 31 and the pre-filtering part 32 depicted in Fig. 3, these parts will hereinbelow be described in detail.

[0022] Fig. 4 illustrates in block form the pre-filter coefficient vector calculation part 31 based on the discussion above. Reference numeral 41 denotes a linear prediction part, 42 a pre-filter deriving coefficient vector correcting part, 43 a pre-filter coefficient vector correcting part, and 44 an error vector generating part. The linear prediction part 41 calculates the forward linear prediction coefficient vector a(k) which satisfies a normal equation $R_p(k)a(k) = [F(k), 0, ..., 0]^T$ and the sum of forward a posteriori prediction-error squares F(k) when the prediction coefficient a(k), the backward linear prediction coefficient vector b(k) which satisfies a normal equation $R_p(k-1)b(k-1) = [0, ..., 0, B(k-1)]^T$ and the sum of backward a posteriori prediction-error squares B(k) when the prediction coefficient b(k) is used. These values can be calculated by methods disclosed in the afore-mentioned literature by J. M. Cioffi et al. The error signal vector generating part 44 stores p error signals e(k), e(k-1), ..., e(k-p+1) and constitutes the error signal vector e(k) of Eq. (11). In the pre-filter deriving coefficient vector correcting part 42, the pre-filter coefficient vector g(k-1), the backward linear prediction coefficient vector b(k-1), the error signal vector e(k-1) and the minimum value of the sum of backward a posteriori prediction-error squares B(k-1) are used to compute the pre-filter deriving coefficient vector f(k-1) on the basis of Eq. (27) or (30). In the pre-filter coefficient vector correcting part 43, the pre-filter deriving vector f(k-1), the error signal vector e(k), the forward linear prediction coefficient vector a(k) and the minimum value of the sum of forward a posteriori prediction-error squares F(k) are used to compute, by Eq. (25) or (29), the vector g(k) that satisfies Eq. (13).

[0023] With the methods mentioned above, the computational complexity involved in the pre-filter coefficient vector calculation part 31 can substantially be reduced from $p^3/6$ to 15p. Since the computational complexity (p-1)L in the pre-filtering part 32 remains unsolved, a large number of operations are still needed when L is large.

[0024] Next, a description will be given of a solution to the above-noted problem by storing approximate values of the estimated transfer function vector $\hat{h}(k+1)$ and averaging pre-filtering coefficients.

[0025] At first, substituting k-1 for k in Eq. (4), the estimated transfer function $\hat{h}(k)$ is expressed as follows:

$$\hat{h}(k) = \hat{h}(k-1) + \mu\delta\hat{h}(k-1) \tag{31}$$

Substituting this in Eq. (4), the estimated transfer function $\hat{h}(k+1)$ is expressed as follows:

$$\hat{h}(k+1) = \hat{h}(k-1) + \mu\delta\hat{h}(k) + \mu\delta h(k-1) \tag{32}$$

Setting k-2, k-3, ... for k in Eq. (4) and repeating the above substitution, the estimated transfer function $\hat{h}(k+1)$ is expressed by

$$\hat{h}(k+1) = \mu\delta\hat{h}(k) + \mu\delta\hat{h}(k-1) + \ldots + \mu\delta\hat{h}(0) \tag{33}$$

In this instance, the estimated initial value $\hat{h}(0)$ is set to 0 and this equation reveals that the estimated transfer function $\hat{h}(k+1)$ is a summation of correcting vectors $\mu\delta\hat{h}(k)$, $\mu\delta\hat{h}(k-1)$, ..., $\mu\delta\hat{h}(0)$ from time 0 (the transfer function estimation starting time) to the current time k.

[0026] The correcting vector $\mu\delta\hat{h}(k)$ is expressed by Eq. (15). Setting k-1, k-2, ..., for k in Eq. (15) and substituting Eq. (15) in Eq. (33), we get

$$\begin{aligned}
\hat{h}(k+1) &= \mu[\{g_1(k)x(k)+g_2(k)x(k-1)+ \ldots + g_p(k)x(k-p+1)\} + \{g_1(k-1)x(k-1)+g_2(k-1)x(k-2) \\
&\quad + \ldots +g_p(k-1)x(k-p)\} + \ldots + \{g_1(0)x(0)+g_2(0)x(-1) + \ldots + g_p(0)x(-p+1)\}] \\
&= \mu[g_1(k)x(k) + \{g_2(k)+g_1(k-1)\}x(k-1) + \{g_3(k)+g_2(k-1)+g_1(k-2)\}x(k-2) + \ldots \\
&\quad + \{g_{p-1}(k)+g_{p-2}(k-1)+ \ldots + g1(k-p+2)\}x(k-p+2) + \{g_p(k)+g_{p-1}(k-1)+ \ldots + g_2(k-p+2) \\
&\quad + g_1(k-p+1)\}x(k-p+1) + \{g_p(k-1)+g_{p-1}(k-2) + \ldots + g_2(k-p+1)+g_1(k-p)\}x(k-p) \\
&\quad + \{g_p(k-2)+g_{p-1}(k-3)+ \ldots + g_2(k-p) + g_1(k-p-1)\}x(k-p-1) + \ldots
\end{aligned} \tag{34}$$

[0027] From this equation the following facts are expected. Firstly, the pre-filter coefficient $g_i(k)$ is calculated at every time k in the pre-filter coefficient calculation part 31 in Fig. 3 and provided to the pre-filtering part 32; in this case, it is expected that the computational complexity would be reduced by smoothing (or averaging) the pre-filter coefficient $g_i(k)$. Secondly, the term + $g_p(k-1)$+ ... and the subsequent terms in Eq. (34) do not involve the pre-filter coefficient $g_i(k)$ which is fed at time k, and hence this portion does not change after time k. By storing these terms as approximate values of the vector $\hat{h}(k+1)$, it is expected that the computational complexity would be reduced accordingly, because no calculations are needed for these terms after time k.

[0028] Next, the above will be expressed by mathematical formulae. The smoothing of the pre-filter coefficient takes place for each corresponding input vector x(k-i). From Eq. (34), the smoothing corresponding to the input vector x(k-1), for example, is $g_2(k)+g_1(k-1)$, and the smoothing corresponding to the input vector x(k-2) is $g_3(k)+g_2(k-1)+g_1(k-2)$. Letting the result of averaging (a smoothing coefficient) corresponding to the input vector x(k-i+1), inclusive of the constant μ, be represented by $s_i(k)$, it is expressed by

$$s_i(k) = \mu \sum_{j=0}^{i-1} g_{i-j}(k-j) \text{ for } 1 \le i \le p \tag{35}$$

$$s_i(k) = \mu \sum_{j=i-p}^{i-1} g_{i-j}(k-j) \text{ for } p < i \tag{36}$$

Eq. (35) is expressed as follows:

$$s_i(k) = \mu \sum_{j=1}^{i-1} g_{i-j}(k-j) + \mu g_i(k) \qquad (37)$$
$$= \mu \sum_{j=0}^{i-2} g_{i-j-1}(k-j-1) + \mu g_i(k)$$
$$= s_{i-1}(k-1) + \mu g_i(k) \qquad \text{for } 2 \le i \le p$$
$$= \mu g_i(k) \qquad \text{for } i = 1$$

[0029] On the other hand, letting the approximate value of the estimated transfer function $\hat{h}(k+1)$ be represented by z(k+1), it is expressed by

$$\begin{aligned} z(k+1) = {} & \{g_p(k)+g_{p-1}(k-1)+ \dots +g_2(k-p+2)+g_1(k-p+1)\}x(k-p+1) \\ & +\{g_p(k-1)+g_{p-1}(k-2)+ \dots +g_2(k-p+1)+g_1(k-p)\}x(k-p) \\ & +\{g_p(k-2)+g_{p-1}(k-3)+ \dots +g_2(k-p)+g_1(k-p-1)\}x(k-p-1) + \dots \end{aligned} \qquad (38)$$

From Eqs. (34), (35) and (38) we have the estimated transfer function $\hat{h}(k+1)$ expressed as follows:

$$\hat{h}(k+1) = z(k+1) + \sum_{i=1}^{p-1} s_i(k) \times (k-i+1) \qquad (39)$$

Moreover, the following relationship holds between the approximate values z(k+1) and z(k).

$$\begin{aligned} z(k+1) &= \{g_p(k)+g_{p-1}(k-1)+ \dots +g_2(k-p+2) +g_1(k-p+1)\}x(k-p+1)+z(k) \\ &= s_p(k)x(k-p+1)+z(k) \end{aligned} \qquad (40)$$

[0030] The estimated value $\hat{y}(k)$ of the output from the unknown system is expressed, from Eqs. (2) and (39), as follows:

$$\hat{y}(k) = x(k)^T \hat{h}(k) \qquad (41)$$
$$= x(k)^T \{z(k) + \sum_{i=1}^{p-1} s_i(k-1)x(k-i)\}$$
$$= x(k)^T z(k) + \sum_{i=1}^{p-1} s_i(k-1)x(k)^T x(k-i)$$
$$= x(k)^T z(k) + s_{p-1}(k-1)^T r_{p-1}(k)$$

In the above, $s_{p-1}(k-1)$ is a smoothing coefficient vector and $r_{p-1}(k)$ is a correlation vector, which are defined by the following equations.

$$s_{p-1}(k-1) = [s_1(k-1), s_2(k-1), \dots, s_{p-1}(k-1)]^T \qquad (42)$$

$$r_{p-1}(k) = [x(k)^T x(k-1), x(k)^T x(k-2), \dots, x(k)^T x(k-p-1)]^T \qquad (43)$$

Since the vector x(k), defined by Eq. (3), is given by

$$x(k) = [x(k), x(k-1), ..., x(k-L+1)]^T \quad (44)$$

the following relationship holds

$$x(k)^T x(k-i) = x(k-1)^T x(k-i-1) - x(k-L)x(k-L-i) + x(k)x(k-i) \quad (45)$$

where i = 1, 2, ..., p-1
and the following equation holds

$$r_{p-1}(k) = r_{p-1}(k-1) - x(k-L)x_{p-1}(k-L) + x(k)x_{p-1}(k) \quad (46)$$

where

$$x_{p-1}(k) = [x(k-1), x(k-2), ..., x(k-p+1)]^T \quad (47)$$

[0031] Next, a description will be given, with reference to Fig. 5, of the storage of the approximate value z(k) of the estimated transfer function $\hat{h}(k)$ and the transfer function estimation procedure which is followed when the pre-filter coefficients are smoothed. At time k, a correlation calculation part 52, which is supplied with the input signal x(k), calculates the correlation vector $r_{p-1}(k)$ by Eq. (46), using the input signal x(k), previous input values x(k-1), ..., x(k-L) and the correlation vector $r_{p-1}(k-1)$ at the immediately preceding time.

[0032] Then, an inner product $s_{p-1}(k-1)^T r_{p-1}(k)$ of the correlation vector $r_{p-1}(k)$ and the smoothing coefficient vector $s_{p-1}(k-1)$ of the pre-filter coefficients is calculated in an inner product calculation part 53. A convolution part 54 performs a convolution $x(k)^T z(k)$ of the stored approximate value z(k) of the estimated transfer function and the input signal. The results of the inner product calculation and the convolution are added together in an addition part 57 to synthesize the estimated value $\hat{y}(k)$ of the unknown system output. These operations correspond to the operation of Eq. (41).

[0033] Following this, the error e(k) between the unknown system output y(k) and the estimated value $\hat{y}(k)$ is obtained by the subtractor 24 shown in Fig. 2 and the pre-filter coefficient $g_i(k)$ is calculated in the pre-filter coefficient vector calculation part 31 shown in Fig. 4.

[0034] After this, the calculated pre-filter coefficients are sent to a pre-filter coefficient smoothing part 51 in Fig. 5, wherein they are smoothed to obtain p smoothing coefficients $s_1(k), s_2(k), ..., s_{p-1}(k), s_p(k)$. This smoothing operation is performed on the basis of Eq. (37). Of the smoothing coefficients, $s_1(k), s_2(k), ..., s_{p-1}(k)$ are supplied, as elements of the smoothing coefficient vector $s_{p-1}(k)$, to the inner product calculation part 53 and an estimated transfer function calculation part 56. The smoothing coefficient $s_p(k)$ is fed to an estimated transfer function approximate value storage part 55.

[0035] The estimated transfer function approximate value storage part 55 updates the approximate value, using the smoothing coefficient $s_p(k)$ and the input signal vector x(k). That is, $s_p(k)x(k-p+1)$ is added to the approximate value z(k) stored until then and the added result is stored as z(k+1). This operation corresponds to the computation Eq. (40).

[0036] Finally, in the estimated transfer calculation part 56 the smoothing coefficient vector $s_{p-1}(k)$ and the input signal vector x(k) are used to calculate Eq. (39) to obtain the estimated transfer function $\hat{h}(k+1)$.

[0037] In the above-described operations, rough estimates of computational complexity involved in the respective parts are as follows:

Correlation calculation part 52:	2p
Inner product calculation part 53:	p
Convolution part 54:	L
Pre-filter coefficient vector calculation part 31:	16p
Pre-filter coefficient smoothing part 51:	p
Linear approximate value storage part 55:	L
Estimated transfer function calculation part 56:	Lp

In the above, p-1 is regarded as nearly equal to p. The entire computational complexity is such as follows:

$$(L+20)p+2L \tag{48}$$

[0038] Now, attention should be paid to the following points. With the conventional transfer function estimation method depicted in Fig. 2, the estimated transfer function $\hat{h}(k+1)$ corresponding to the transfer function is calculated at every time and is used to synthesize the estimated value $\hat{y}(k)$ of the unknown system output. In contrast thereto, according to the present embodiment which utilizes the approximate value z(k) of the estimated transfer function, the estimated value $\hat{y}(k)$ of the unknown system output can be synthesized without the need of calculating, directly, the estimated transfer function $\hat{h}(k+1)$ as shown in Fig. 5. Once the estimated value $\hat{y}(k)$ is obtained, the above-described operations can be performed. Accordingly, there is no need of calculating the estimated transfer function at every time in the cases where the estimated value $\hat{h}(k+1)$ of the transfer function, obtained as the result of the long-time estimating operation, is needed and where the estimated value $\hat{y}(k)$ of the unknown system output is needed rather than the estimated result of the transfer function (for example, in the case of the estimation of characteristics of a time-invariant system or in an acoustic echo canceller).

[0039] On this account, the overall computational complexity at every time, except the computational complexity in the estimated transfer function calculation part 56, is as follows:

$$20p + 2L \tag{49}$$

As referred to previously with respect to the prior art, the number of operations needed in the past is about 12000 when the tap number L of the filter is 500 and the projection order p is 20. From Eq. (49), however, the number of operations in the present invention is 1380; that is, the computational complexity is reduced to about 1/8 that in the prior art.

[0040] As described above, the present invention permits substantial reduction of the computational complexity involved in the conventional projection scheme by the recursive synthesis of the pre-filter coefficient, storage of approximate values of the estimated transfer function vector and smoothing of the pre-filtering coefficients.

[0041] While in the above the step size μ has been described to be handled as a scalar, it may also be provided as μA by use of a diagonal matrix A. For example, in the case where the energy of an impulse response of an unknown system decays exponentially, the transfer function estimation speed may sometimes be increased by arranging the elements of the diagonal matrix A as a progression which decays exponentially, as shown below.

$$A = \mathrm{diag}(\alpha, \alpha\gamma, \alpha\gamma^2, \ldots, \alpha\gamma^{L-1}), (0 < \gamma < 1) \tag{50}$$

In this instance, the input to the linear prediction part is multiplied by a ratio $\gamma^{1/2}$ and $R_p(k)$ Eq. (14) is redefined by the following equation.

$$R_p(k) = \begin{bmatrix} \boldsymbol{x}(k)^T \\ \boldsymbol{x}(k-1)^T \\ \vdots \\ \boldsymbol{x}(k-p+1)^T \end{bmatrix} A \left[\boldsymbol{x}(k)\boldsymbol{x}(k-1) \ \ldots \ \boldsymbol{x}(k-p+1)\right] \tag{51}$$

This correction permits the application of the present invention described above.

[0042] Next, a description will be given of examples of application of the transfer function estimating device of the present invention.

[0043] A first example of application is the measurements of transfer functions of pieces of acoustic equipment. Fig. 6 illustrates a system for measuring the transfer function from a loudspeaker to a microphone. In Fig. 6, reference numeral 121 denotes a loudspeaker, 122 a microphone and 11 a transfer function estimating device. The output signal y(k) of the microphone 122 is a signal that has the characteristics of the loudspeaker 121 added to the input signal x(k). Regarding the loudspeaker 121 (including an acoustic path and the microphone as well) as an unknown system, the illustrated system is the same as that shown in Fig. 1, and by connecting the transfer function estimating device 11 of the present invention to the input of the loudspeaker 121 and the output of the microphone 122 as shown in Fig. 6, the transfer function h(k) of the loudspeaker can be estimated as the filter coefficient $\hat{h}(k)$ of the FIR filter. The measurement is usually performed in an anechoic chamber to avoid the influence of room acoustical characteristics.

[0044] A second example is an acoustic echo canceller which prevents howling or echo in a loudspeaker communication system such as a TV conference system or visual telephone. Fig. 7 is a diagram for explaining the acoustic

echo canceller. In Fig. 7, reference numeral 121 denotes a receiving loudspeaker, 122 a transmitting microphone, 123 a room acoustic system and 20 an estimated echo generating part which is identical in construction with the estimated signal generating part 20 embodying the present invention in Fig. 2. The transfer function estimating device 11 of this embodiment operates as an acoustic echo canceller. In a hands-free communication system using the loudspeaker and the microphone, the other party's voice emanating from the receiving loudspeaker 121 is received by the transmitting loudspeaker via the room acoustic system of the transfer function h(k). The received signal y(k) is returned to the other party and reproduced there. At the other party side, the transmitted voice is sent back and reproduced; this phenomenon is called an acoustic echo and disturbs comfortable communication.

[0045] The estimated echo generating part 20 of the acoustic echo canceller 11 estimates the room acoustical characteristics h(k) including characteristics of the loudspeaker and generates an estimated echo signal $\hat{y}$ (k) based on the characteristics $\hat{h}$ (k) estimated as those of the input signal x(k). The subtractor 24 subtracts the estimated echo signal $\hat{y}$ (k) from the received signal y(k). When the estimation is performed well, the echo canceller 11 operates so that it minimizes the power of the error signal e(k) and hence makes the estimated echo signal $\hat{y}$ (k) nearly equal to the received signal y(k), substantially reducing the acoustic echo.

[0046] Comparison of the Fig. 7 system with the Fig. 2 system reveals that the transfer function estimating device of the present invention can be used directly as an echo canceller. The room acoustic system 123 in Fig. 7 corresponds to the unknown system 12 in Fig. 2 and the estimated echo $\hat{y}$ (k) and the transmitted signal e(k) in Fig. 7 correspond to the output y(k) from the convolution part 23 and the error signal e(k) in Fig. 2, respectively.

[0047] A third example is noise control. Fig. 8 illustrates the principles of noise control. In Fig. 8, reference numeral 131 denotes a noise source, 132 a noise transfer path expressed by the transfer function h(k), 133 a microphone for observation, 134 a noise monitor microphone, 20 an estimated noise generating part, 136 a phase inverter and 137 a loudspeaker. The purpose of noise control is to cancel noise that is observed via the noise transfer path of the transfer characteristics h(k) from the noise source 131, by generating negative estimated noise (letting a sound pressure be represented by y(k), a sound pressure represented by -y(k) is called a negative sound with respect to y(k)) from the loudspeaker 137.

[0048] To attain this object, the transfer characteristics h(k) of the noise transfer path 132 is estimated in the estimated noise generating part 20 of the same construction as that of the estimated signal generating part 20 in Fig. 2. That is, noise is detected by the monitor microphone 134 in the vicinity of the noise source 131 and provided as the input signal x(k) to the estimated noise generating part 20, which generates an estimated value $\hat{y}$ (k) of the noise signal y(k) at the observation point (i.e. the microphone 133). The estimated noise $\hat{y}$ (k) is polarity inverted by the phase inverter 136 into a signal -y(k). Assuming, for the sake of brevity, that the loudspeaker characteristics are negligible, the signal -$\hat{y}$ (k) from the loudspeaker 137 is a combination of the noise signal y(k) and the sound pressure in the microphone 133 at the observation point and the error signal e(k) is provided at the output of the microphone 133. In this instance, when the noise transfer path is estimated well, the signal $\hat{y}$ (k) becomes similar to the noise y(k) from the noise source 131 and the noise y(k) is cancelled by the combined sound pressure -$\hat{y}$ (k) from the loudspeaker 137.

[0049] The microphone 133 in Fig. 8 corresponds to the subtractor 24 in Fig. 2; accordingly, the systems of Figs. 8 and 2 differ only in whether the error signal generated outside the estimating device 11 is provided thereto or the error signal is calculated in the estimating device 11. The principles of the present invention are applicable to the noise control device of Fig. 8.

[0050] In the measurement of the transfer characteristics of the loudspeaker shown in Fig. 6, the loudspeaker characteristics do not change with time; hence, there is no need of learning measured results (transfer characteristics) in the course of measurement and the transfer characteristics need only to be made known as the result of measurement conducted after a certain period of time. In the acoustic echo canceller shown in Fig. 7, the room transfer function varies with time in response to movement of audiences or opening and closing of a door, for instance. As is evident from Fig. 7, however, the purpose of the acoustic echo canceller is attained by obtaining the estimated value $\hat{y}$ (k) of the output y(k) of the room transfer function (the unknown system). Therefore, the estimated value $\hat{h}$ (k) of the room transfer function itself is unnecessary in the acoustic echo canceller. Also in the noise control device of Fig. 8, its purpose can be attained by obtaining the estimated value $\hat{y}$ (k) of the unknown system and no estimated value of the room transfer function itself is needed.

[0051] As will be appreciated from the above, the transfer function estimating device of the present invention, when applied as described above, permits substantial reduction of the computational complexity as compared with the prior art. Furthermore, the present invention has a basic feature of minimizing the power of the error signal e(k) in such a system configuration as shown in Fig. 2. Thus, the present invention is applicable to all instances in which problems to be solved can be modelled as the error signal power minimizing problem shown in Fig. 2.

[0052] In the embodiments described above, it will be effective to select the elements of the diagonal matrix A of Eq. (50) so that they provide the same attenuation as the room reverberation.

[0053] Finally, a description will be given of experimental results of the acoustic echo canceller according to the present invention. Fig. 9 shows what are called learning curves, the ordinate representing echo return loss enhance-

ment (hereinafter referred to as ERLE) and the abscissa representing time. As the estimation of the room transfer function proceeds with the lapse of time, the ERLE increases. In Fig. 9, the curve 211 is a learning curve in the case of the projection order p being 2, the curve 212 a learning curve in the case of the projection order p being 8 and the curve 213 a learning curve in the case of the projection order p being 32. It will be understood, from Fig. 9, that the larger the projection order, the higher the convergence speed (the ERLE increases in a short period of time).

[0054] Fig. 10 is a graph showing the relationship between the projection order p and the required computational complexity, the abscissa representing the projection order p and the ordinate the number of multiply-add operations (including add operations as well). In Fig. 10, the curve 221 indicates the computational complexity in the prior art and the curve 222 the computational complexity when the present invention was used. The tap number L of the FIR filter for use in the estimation of the transfer function was 500. From Fig. 10 it will be seen that the computational complexity involved in the present invention is much smaller than in the case of the prior art when the projection order p is large.

[0055] As described above, the present invention permits substantial reduction of the number of operations necessary for the estimation of the transfer function or output of an unknown system by the projection scheme. In concrete terms, letting the tap length of the FIR filter which represents the unknown system and the projection order be represented by L and p, respectively, the prior art requires multiply-add operations $p^3/6+(p+1)L$, that is, the computational complexity is proportional to p^3, whereas according to the present invention the computation complexity can be reduced down to 20p+2L.

[0056] Such a reduction of computational complexity allows corresponding reduction in the scale of hardware, and hence significantly contributes to the downsizing of the device and reduction of its cost. Moreover, when the hardware is held on the same scale, the tap length L and the projection order can be chosen larger than in the past -- this speeds up the estimating operation and increases the estimation accuracy. Besides, when the present invention is embodied by a computer, the operation time can be greatly shortened.

Claims

1. A method of adaptively estimating the transfer function of an unknown technical system by use of a projection algorithm, said method comprising:

   (i) applying an input physical signal x(k) to said unknown system (12), k representing a discrete time;
   (ii) detecting an output physical signal y(k) from said unknown system (12);
   (iii) estimating the transfer function using said input signal x(k) and said output signal y(k) and generating an output signal $\hat{y}(k)$ of an estimated system having said estimated transfer function $\hat{h}(k)$, wherein said estimated system is implemented as a digital filter of tap number L;
   (iv) obtaining, as an error signal $e(k)=y(k)-\hat{y}(k)$, the difference between the output signal $\hat{y}(k)$ of said estimated system and the output signal y(k) of said unknown system (12);
   (v) calculating a correcting vector $\delta\hat{h}(k)$ for the estimated transfer function from input signal vectors x(k), x(k-1), ..., x(k-p + 1) and error signals e(k), e(k-1), ... , e(k-p + 1), where p is an integer equal to or greater than 2 and x(k) is defined as $x(k)=[x(k), x(k-1),..., x(k-L+ 1)]^T$; and
   (vi) using said correcting vector $\delta\hat{h}(k)$ and a predetermined correcting step size μ to repeatedly correct said estimated transfer function $\hat{h}(k)$ at each time k by the following equation:

$$\hat{h}(k+1) = \hat{h}(k) + \mu\delta\hat{h}(k)$$

   so that said error signal e(k) approaches zero;
   characterized in that step (v) comprises the steps of:

   (v-1) calculating a pre-filter coefficient vector g(k) by solving the following simultaneous linear equation with p unknowns:

$$R_p(k)g(k)=e(k),$$

   wherein e(k) represents the vector of said error signal e(k) and is defined as

$$e(k)=[e(k), (1-\mu)e(k-1),..., (1-\mu)^{p-1}e(k-p+1)]^T, \text{ and}$$

   $R_p(k)$ is a covariance matrix of said input signal x(k) defined as

$$R_p(k) = [x(k), x(k-1),..., x(k-p + 1)]^T [x(k), x(k-1),..., x(k-p + 1)]; \text{ and}$$

(v-2) using said pre-filter coefficient vector 9(k) to calculate said correcting vector $\delta\hat{h}(k)$ by the following equation:

$$\delta\hat{h}(k) = [x(k), x(k-1), ..., x(k-p+1)]g(k),$$

wherein x(k) represents the vector of said input signal x(k);
wherein step (v-1) comprises the steps of:
(v-1-a) calculating a forward linear prediction coefficient vector a(k) of said input signal x(k), the sum of its a posteriori prediction-error squares F(k), a backward linear prediction coefficient vector b(k) of said input signal x(k) and the sum of its a posteriori prediction-error squares B(k), and
(v-1-b) letting a pre-filter deriving coefficient vector be represented by f(k), obtaining said pre-filter coefficient vector g(k) by a recursion formula composed of the following first and second equations:

$$\mathbf{g}(k) = (1-\mu)\begin{bmatrix} 0 \\ \mathbf{f}(k-1) \end{bmatrix} + \frac{\mathbf{a}(k)^T e(k)}{F(k)}\mathbf{a}(k)$$

$$\begin{bmatrix} \mathbf{f}(k-1) \\ 0 \end{bmatrix} = \mathbf{g}(k-1) - \frac{\mathbf{b}(k-1)^T e(k-1)}{B(k-1)}\mathbf{b}(k-1)$$

2. A method of adaptively estimating the transfer function of an unknown technical system by use of a projection algorithm, said method comprising:

(i) applying an input physical signal x(k) to said unknown system (12), k representing a discrete time;
(ii) detecting an output physical signal y(k) from said unknown system (12);
(iii) estimating the transfer function using said input signal x(k) and said output signal y(k) and generating an output signal $\hat{y}(k)$ of an estimated system having said estimated transfer function $\hat{h}(k)$, wherein said estimated system is implemented as a digital filter of tap number L;
(iv) obtaining, as an error signal $e(k)=y(k)-\hat{y}(k)$, the difference between the output signal $\hat{y}(k)$ of said estimated system and the output signal y(k) of said unknown system (12);
(v) repeating the calculation of the estimated transfer function at each time k so that said error signal e(k) approaches zero;
characterized in that
step (v) comprises steps of:

(v-1) calculating a pre-filter coefficient vector g(k) by solving the following simultaneous linear equation with p unknowns:

$$R_p(k)g(k)=e(k),$$

wherein e(k) represents the vector of said error signal e(k) and is defined as .

$$e(k)=[e(k), (1-\mu)e(k-1),..., (1-\mu)^{p-1}e(k-p+1)]^T, \text{ and}$$

$R_p(k)$ is a covariance matrix of said input signal x(k) defined as

$$R_p(k) = [x(k), x(k-1),..., x(k-p+1)]^T [x(k), x(k-1),..., x(k-p+1)];$$

(v-2) smoothing pre-filter coefficients $g_i(k)$ which are elements of said pre-filter coefficient vector g(k) by the following equations:

$$s_i(k)=s_{i-1}(k-1)+\mu g_i(k) \qquad \text{for } 2\le i\le p$$
$$=\mu g_1(k) \qquad \text{for } i=1$$

to obtain a smoothing coefficient $s_i(k)$;
(v-3) obtaining an approximate estimated transfer function z(k + 1) as an approximation of said estimated transfer function using the smoothing coefficient $s_p(k)$ based on the following equation

$$z(k+1)=z(k)+s_p(k)x(k-p+1);$$

(v-4) calculating the convolution $x(k)^T z(k)$ of said approximate estimated transfer function z(k) with an input signal vector x(k) defined as $x(k)=[x(k), x(k-1),..., x(k-L+1)]^T$;
(v-5) calculating the inner product $s_{p-1}(k-1)^T r_{p-1}(k)$, setting the vector of said smoothing coefficient $s_i(k)$ and a correlation vector $r_{p-1}(k)$ of said input signal to

$$s_{p-1}(k-1)=[s_1(k-1), s_2(k-1), ..., s_{p-1}(k-1)]^T,$$

and

$$r_{p-1}(k)=[x(k)^T x(k-1), x(k)^T x(k-2), ..., x(k)^T x(k-p+1)]^T,$$

respectively; and
(v-6) obtaining the sum of said convolution result $x(k)^T z(k)$ and said inner product as said estimated output signal $\hat{y}(k)$.

3. The method of claim 2, wherein step (v-1) comprises:

(a) calculating a forward linear prediction coefficient vector a(k) of said input signal x(k), the sum of its a posteriori prediction-error squares F(k), a backward linear prediction coefficient vector b(k) of said input signal x(k) and the sum of its a posteriori prediction-error squares B(k), and
(b) letting a pre-filter deriving coefficient vector be represented by f(k), obtaining said pre-filter coefficient vector g(k) by a recursion formula composed of the following first and second equations:

$$\mathbf{g}(k)=(1-\mu)\begin{bmatrix}0\\ \mathbf{f}(k-1)\end{bmatrix}+\frac{\mathbf{a}(k)^T\mathbf{e}(k)}{F(k)}\mathbf{a}(k)$$
$$\begin{bmatrix}\mathbf{f}(k-1)\\ 0\end{bmatrix}=\mathbf{g}(k-1)-\frac{\mathbf{b}(k-1)^T\mathbf{e}(k-1)}{B(k-1)}\mathbf{b}(k-1)$$

4. The method of claim 1 or 3, wherein, letting the last element of said pre-filter coefficient vector g(k-1) be represented by $g_p(k-1)$, said pre-filter coefficient vector g(k) is calculated using the following equation which is a modified version of said second equation:

$$\begin{bmatrix}\mathbf{f}(k-1)\\ 0\end{bmatrix}=\mathbf{g}(k-1)-g_p(k-1)\mathbf{b}(k-1)$$

5. The method of claim 1 or 3, wherein, letting predetermined two non-negative numbers be represented by $\delta_F(k)$ and $\delta_B(k-1)$, said pre-filter coefficient vector g(k) is calculated by the following equations which are modified versions of

said first and second equations:

$$\mathbf{g}(k) = (1-\mu)\begin{bmatrix} 0 \\ \mathbf{f}(k-1) \end{bmatrix} + \frac{\mathbf{a}(k)^T e(k)}{F(k) + \delta_F(k)}\mathbf{a}(k)$$

$$\begin{bmatrix} \mathbf{f}(k-1) \\ 0 \end{bmatrix} = \mathbf{g}(k-1) - \frac{\mathbf{b}(k-1)^T e(k-1)}{B(k-1) + \delta_B(k-1)}\mathbf{b}(k-1)$$

6. The method of claim 2 or 3 wherein step (v-5) comprises calculating said correlation vector $r_{p-1}(k)$ of said input signal x(k) by

$$r_{p-1}(k) = r_{p-1}(k-1) - x(k-L)x_{p-1}(k-L) + x(k)x_{p-1}(k)$$

where $x_{p-1}(k)$ represents the vector of said input signal x(k) expressed as:

$$x_{p-1}(k) = [x(k-1), x(k-2), ..., x(k-p+1)]^T.$$

7. A device for adaptive estimation of the transfer function of an unknown technical system by use of a projection algorithm, said device comprising:

means for applying an input physical signal x(k) to said unknown system (121, k representing a discrete time;
means for detecting an output physical signal y(k) from said unknown system (12);
means (11) for estimating the transfer function using said input signal x(k) and said output signal y(k) and generating an output signal $\hat{y}(k)$ of an estimated system (23) having said estimated transfer function $\hat{h}(k)$, wherein said estimated system is implemented as a digital filter of tap number L;
means (24) for obtaining, as an error signal $e(k)=y(k)-\hat{y}(k)$, the difference between the output signal $\hat{y}(k)$ of said estimated system and the output signal y(k) of said unknown system (12);
means (21) for calculating a correcting vector $\delta\hat{h}(k)$ for the estimated transfer function from input signal vectors x(k), x(k-1), ... , x(k-p+1) and error signals e(k), e(k-1), ... , e(k-p+1), where p is an integer equal to or greater than 2 and x(k) is defined as $x(k)=[x(k), x(k-1),..., x(k-L+1)]^T$; and
means (22) for repeatedly correcting, at each time k using said correcting vector $\delta\hat{h}(k)$ and a predetermined correcting step size μ, said estimated transfer function $\hat{h}(k)$ by the following equation:

$$\hat{h}(k+1) = \hat{h}(k) + \mu\delta\hat{h}(k)$$

so that said error signal e(k) approaches zero;
characterized in that said means (21) for calculating the correcting vector $\delta\hat{h}(k)$ comprises:

- means (31) for calculating a pre-filter coefficient vector g(k) by solving the following simultaneous linear equation with p unknowns:

$$R_p(k)g(k)=e(k),$$

wherein e(k) represents the vector of said error signal elk) and is defined as

$$e(k)=[e(k), (1-\mu)e(k-1),..., (1-\mu)^{p-1}e(k-p+1)]^T, \text{ and}$$

$R_p(k)$ is a covariance matrix of said input signal x(k) defined as

$$R_p(k)=[x(k), x(k-1),..., x(k-p+1)]^T\,[x(k), x(k-1),..., x(k-p+1)]; \text{ and}$$

- means (32) for calculating, using said pre-filter coefficient vector g(k), said correcting vector $\delta\hat{h}(k)$ by the following equation:

$$\delta\hat{h}(k) = [x(k), x(k-1), ..., x(k-p+1)]g(k),$$

where x(k) represents the vector of said input signal x(k); and

said means (31) for calculating said pre-filter coefficient vector g(k) comprises:

- means (41) for calculating a forward linear prediction coefficient vector a(k) of said input signal x(k), the sum of its a posteriori prediction-error squares F(k), a backward linear prediction coefficient vector b(k) of said input signal x(k) and the sum of its a posteriori prediction-error squares B(k), and
- means (42, 43, 44) for obtaining said pre-filter coefficient vector g(k) by a recursion formula composed of the following first and second equations:

$$g(k) = (1-\mu)\begin{bmatrix} 0 \\ f(k-1) \end{bmatrix} + \frac{a(k)^T e(k)}{F(k)} a(k)$$

$$\begin{bmatrix} f(k-1) \\ 0 \end{bmatrix} = g(k-1) - \frac{b(k-1)^T e(k-1)}{B(k-1)} b(k-1)$$

where f(k) represents a pre-filter deriving coefficient vector.

8. A device for adaptive estimation of the transfer function of an unknown technical system by use of a projection algorithm, said device comprising:

means for applying an input physical signal x(k) to said unknown system (12), k representing a discrete time;
means for detecting an output technical signal g(k) from said unknown system (12);
means (11) for estimating the transfer function using said input signal x(k) and said output signal y(k) and generating an output signal $\hat{y}$ (k) of an estimated system (23) having said estimated transfer function $\hat{h}$ (k), wherein said estimated system is implemented as a digital filter of tap number L;
means (24) for obtaining, as an error signal e(k) = y(k)-$\hat{y}$(k) , the difference between the output signal $\hat{y}$ (k) of said estimated system and the output signal y(k) of said unknown system (12); and
means (20) for calculating the estimated transfer function at each time k so that said error signal e(k) approaches zero;
characterized in that said means (20) for calculating the estimated transfer function comprises:

- means (31) for calculating a pre-filter coefficient vector g(k) by solving the following simultaneous linear equation with p unknowns:

$$R_p(k)g(k) = e(k),$$

wherein e(k) represents the vector of said error signal e(k) and is defined as

$$e(k) = [e(k), (1-\mu)e(k-1), ..., (1-\mu)^{p-1} e(k-p+1)]^T, \text{ and}$$

$R_p(k)$ is a covariance matrix of said input signal x(k) defined as

$$R_p(k) = [x(k), x(k-1), ..., x(k-p+1)]^T [x(k), x(k-1), ..., x(k-p+1)];$$

- means (51) for smoothing pre-filter coefficients $g_i(k)$, which are elements of said pre-filter coefficient vector g(k), by the following equations:

$$s_i(k) = s_{i-1}(k-1) + \mu g_i(k) \quad \text{for } 2 \le i \le p$$
$$= \mu g_1(k) \quad \text{for } i=1$$

to obtain a smoothing coefficient $s_i(k)$; and

- means (55) for obtaining an approximate estimated transfer function z(k + 1) as an approximation of said estimated transfer function using the smoothing coefficient $s_p(k)$ based on the following equation

$$z(k+1) = z(k) + s_p(k)x(k-p+1); \text{ and}$$

said estimated system (23) comprises:

- means (54) for calculating the convolution $x(k)^T z(k)$ of said approximate estimated transfer function z(k) with an input signal vector x(k) defined as $x(k)=[x(k), x(k-1),..., x(k-L+1)]^T$;
- means (53) for calculating the inner product $s_{p-1}(k-1)^T r_{p-1}(k)$, setting the vector of said smoothing coefficient $s_i(k)$ and a correlation vector $r_{p-1}(k)$ of said input signal to

$$s_{p-1}(k-1)=[s_1(k-1), s_2(k-1), ..., s_{p-1}(k-1)]^T$$

and

$$r_{p-1}(k)=[x(k)^T x(k-1), x(k)^T x(k-2), ..., x(k)^T x(k-p+1)]^T$$

respectively; and

- means (57) for obtaining the sum of said convolution result $x(k)^T z(k)$ and said inner product as said estimated output signal $\hat{y}(k)$.

9. The device of claim 8, wherein said means (31) for solving said simultaneous linear equation with p unknowns comprises:

means (41) for calculating a forward linear prediction coefficient vector a(k) of said input signal x(k), the sum of its a posteriori prediction-error squares F(k), a backward linear prediction coefficient vector b(k) of said input signal x(k) and the sum of its a posteriori prediction-error squares B(k), and
means (42, 43, 44) for obtaining said pre-filter coefficient vector g(k) by a recursion formula composed of the following first and second equations:

$$g(k) = (1-\mu)\begin{bmatrix} 0 \\ f(k-1) \end{bmatrix} + \frac{a(k)^T e(k)}{F(k)} a(k)$$

$$\begin{bmatrix} f(k-1) \\ 0 \end{bmatrix} = g(k-1) - \frac{b(k-1)^T e(k-1)}{B(k-1)} b(k-1)$$

where f(k) represents a pre-filter deriving coefficient vector.

10. The device of claim 8 or 9, wherein said inner product calculating means includes correlation calculating means for calculating said correlation vector $r_{p-1}(k)$ of said input signal x(k) by

$$r_{p-1}(k) = r_{p-1}(k-1) - x(k-L)x_{p-1}(k-L) + x(k)x_{p-1}(k),$$

where $x_{p-1}(k)$ represents the vector of said input signal x(k) expressed as:

$$x_{p-1}(k) = [x(k-1), x(k-2), ..., x(k-p+1)]^T.$$

11. The device of claim 7 or 9, wherein said pre-filter deriving coefficient vector correcting means is means which, letting the last element of said pre-filter coefficient vector g(k-1) be represented by $g_p(k-1)$, calculates said pre-filter coefficient vector g(k) by the following equation which is a modified version of said second equation:

$$\begin{bmatrix} f(k-1) \\ 0 \end{bmatrix} = g(k-1) - g_p(k-1)b(k-1)$$

12. The device of claim 7 or 9, wherein, letting predetermined non-negative numbers be represented by $\delta_F(k)$ and $\delta_B(k-1)$, respectively, said pre-filter coefficient vector correcting means and said pre-filter deriving coefficient vector correcting means calculate said pre-filter coefficient vector g(k) by the following equations, respectively, which are modified versions of said first and second equations:

$$g(k) = (1-\mu)\begin{bmatrix} 0 \\ f(k-1) \end{bmatrix} + \frac{a(k)^T e(k)}{F(k)+\delta_F(k)} a(k)$$

$$\begin{bmatrix} f(k-1) \\ 0 \end{bmatrix} = g(k-1) - \frac{b(k-1)^T e(k-1)}{B(k-1)+\delta_B(k-1)} b(k-1)$$

13. The device of claim 7 or 10, further comprising subtracting means for providing, as said error signal e(k), the difference $y(k)-\hat{y}(k)$ between said output signal y(k) of said unknown system (12) and said estimated signal $\hat{y}(k)$ which is the output of said estimated system.

Patentansprüche

1. Verfahren zum adaptiven Schätzen der Übertragungsfunktion eines unbekannten technischen Systems durch Verwendung eines Projektionsalgorithmus, wobei das Verfahren umfaßt:

(i) Anlegen eines physikalischen Eingangssignals x(k) an das unbekannte System (12), wobei k eine diskrete Zeit repräsentiert;
(ii) Erfassen eines physikalischen Ausgangssignals y(k) aus dem unbekannten System (12);
(iii) Schätzen der Übertragungsfunktion unter Verwendung des Eingangssignals x(k) und des Ausgangssignals y(k) und Erzeugen eines Ausgangssignals $\hat{y}(k)$ eines Schätzsystems mit der geschätzten Übertragungsfunktion $\hat{h}(k)$, wobei das Schätzsystem als ein digitales Filter mit einer Anzahl L an Anzapfungen implementiert ist;
(iv) Ermitteln, als ein Fehlersignal $e(k) = y(k)-\hat{y}(k)$, der Differenz zwischen dem Ausgangssignal $\hat{y}(k)$ des Schätzsystems und dem Ausgangssignal y(k) des unbekannten Systems (12);
(v) Berechnen eines Korrigiervektors $\delta\hat{h}(k)$ für die geschätzte Übertragungsfunktion aus Eingangssignalvektoren x(k), x(k-1), ..., x(k-p + 1) und Fehlersignalen e(k), e(k-1), ..., e(k-p + 1), wobei p eine ganze Zahl größer oder gleich 2 ist und x(k) definiert ist als $x(k) = [x(k), x(k-1), ..., x(k-L + 1)]^T$; und
(vi) Verwenden des Korrigiervektors $\delta\hat{h}(k)$ und einer vorbestimmten Korrigierschrittgröße μ, um wiederholt die geschätzte Übertragungsfunktion $\hat{h}(k)$ zu jedem Zeitpunkt k durch die folgende Gleichung zu korrigieren:

$$\hat{h}(k+1)=\hat{h}(k)+\mu\delta\hat{h}(k)$$

so daß sich das Fehlersignal e(k) Null annähert;
dadurch gekennzeichnet, daß Schritt (v) folgende Schritte umfaßt:

(v-1) Berechnen eines Vorfilterkoeffizientenvektors g(k) durch Lösen der folgenden simultanen linearen Gleichung mit p Unbekannten:

$$R_p(k)g(k)=e(k),$$

wobei e(k) den Vektor des Fehlersignals e(k) repräsentiert und definiert ist als

$$e(k)=[e(k), (1-\mu)e(k-1), ..., (1-\mu)^{p-1}e(k-p+1)]^T, \text{ und}$$

$R_p(k)$ eine Kovarianzmatrix des Eingangssignals x(k) ist und definiert ist als

$$R_p(k)=[x(k), x(k-1), ..., x(k-p + 1)]^T [x(k), x(k-1), ..., x(k-p + 1)]; \text{ und}$$

(v-2) Verwenden des Vorfilterkoeffizientenvektors g(k) zum Berechnen des Korrigiervektors $\delta\hat{h}(k)$ durch die folgende Gleichung:

$$\delta\hat{h}(k)=[x(k), x(k-1), ..., x(k-p+1)]\, g(k),$$

wobei x(k) den Vektor des Eingangssignals x(k) repräsentiert;
wobei Schritt (v-1) folgende Schritte umfaßt:
(v-1-a) Berechnen eines Vorwärtslinearvorhersagekoeffizientenvektors a(k) des Eingangssignals x(k), der Summe seiner a-posteriori-Vorhersagefehlerquadrate F(k), eines Rückwärtslinearvorhersagekoeffizientenvektors b(k) des Eingangssignals x(k) und der Summe seiner a-posteriori-Vorhersagefehlerquadrate B(k), und
(v-1-b) wenn ein Vorfilterableitungskoeffizientenvektor durch f(k) repräsentiert sei, Ermitteln des Vorfilterkoeffizientenvektors g(k) durch eine Rekursionsformel, die sich aus der folgenden ersten und zweiten Gleichung zusammensetzt:

$$g(k) = (1-\mu)\begin{bmatrix} 0 \\ f(k-1) \end{bmatrix} + \frac{a(k)^T e(k)}{F(k)} a(k)$$

$$\begin{bmatrix} f(k-1) \\ 0 \end{bmatrix} = g(k-1) - \frac{b(k-1)^T e(k-1)}{B(k-1)} b(k-1)$$

2. Verfahren zum adaptiven Schätzen der Übertragungsfunktion eines unbekannten technischen Systems durch Verwendung eines Projektionsalgorithmus, wobei das Verfahren umfaßt:

(i) Anlegen eines physikalischen Eingangssignals x(k) an das unbekannte System (12), wobei k eine diskrete Zeit repräsentiert;
(ii) Erfassen eines physikalischen Ausgangssignals y(k) aus dem unbekannten System (12);
(iii) Schätzen der Übertragungsfunktion unter Verwendung des Eingangssignals x(k) und des Ausgangssignals y(k) und Erzeugen eines Ausgangssignals $\hat{y}(k)$ eines Schätzsystems mit der geschätzten Übertragungsfunktion $\hat{h}(k)$, wobei das Schätzsystem als ein digitales Filter mit einer Anzahl L an Anzapfungen implementiert ist;
(iv) Ermitteln, als ein Fehlersignal $e(k)=y(k)-\hat{y}(k)$, der Differenz zwischen dem Ausgangssignal $\hat{y}(k)$ des Schätzsystems und dem Ausgangssignal y(k) des unbekannten Systems (12);
(v) Wiederholen der Berechnung der geschätzten Übertragungsfunktion zu jedem Zeitpunkt k so, daß sich das Fehlersignal e(k) Null annähert;
dadurch gekennzeichnet, daß
Schritt (v) folgende Schritte umfaßt:

(v-1) Berechnen eines Vorfilterkoeffizientenvektors g(k) durch Lösen der folgenden simultanen linearen Gleichung mit p Unbekannten:

$$R_p(k)g(k)=e(k),$$

wobei e(k) den Vektor des Fehlersignals e(k) repräsentiert und definiert ist als

$$e(k) =[e(k), (1-\mu)e(k-1), ..., (1-\mu)^{p-1}e(k-p+1)]^T, \text{ und}$$

$R_p(k)$ eine Kovarianzmatrix des Eingangssignals x(k) ist und definiert ist als

$$R_p(k) = [x(k), x(k-1), ..., x(k-p+1)]^T [x(k), x(k-1), ..., x(k-p+1)];$$

(v-2) Glätten von Vorfilterkoeffizienten g(k), die Elemente des Vorfilterkoeffizientenvektors g(k) sind, durch die folgenden Gleichungen:

$$s_i(k)=s_{i-1}(k-1)+\mu g_i(k) \qquad \text{für } 2\le i\le p$$
$$=\mu g_1(k) \qquad \text{für } i=1$$

um einen Glättungskoeffizienten $s_i(k)$ zu ermitteln;

(v-3) Ermitteln einer ungefähren geschätzten Übertragungsfunktion z(k + 1) als einer Näherung der geschätzten Übertragungsfunktion unter Verwendung des Glättungskoeffizienten $s_p(k)$ auf der Basis der folgenden Gleichung

$$z(k+1)=z(k)+s_p(k)x(k-p+1);$$

(v-4) Berechnen der Faltung $x(k)^T z(k)$ der ungefähren geschätzten Übertragungsfunktion z(k) mit einem Eingangssignalvektor x(k), der definiert ist als $x(k) = [x(k), x(k-1), ..., x(k-L+1)]^T$;

(v-5) Berechnen des inneren Produkts $s_{p-1}(k-1)^T r_{p-1}(k)$, wobei der Vektor des Glättungskoeffizienten $s_i(k)$ und ein Korrelationsvektor $r_{p-1}(k)$ des Eingangssignals auf

$$s_{p-1}(k-1)=[s_1(k-1), s_2(k-1), ..., s_{p-1}(k-1)]^T,$$

bzw.

$$r_{p-1}(k)=[x(k)^T x(k-1), x(k)^T x(k-2), ..., x(k)^T x(k-p+1)]^T,$$

gesetzt werden; und

(v-6) Ermitteln der Summe des Faltungsergebnisses $x(k)^T z(k)$ und des inneren Produkts als das geschätzte Ausgangssignal $\hat{y}(k)$.

3. Verfahren nach Anspruch 2, bei dem Schritt (v-1) umfaßt:

(a) Berechnen eines Vorwärtslinearvorhersagekoeffizientenvektors a(k) des Eingangssignals x(k), der Summe seiner a-posteriori-Vorhersagefehlerquadrate F(k), eines Rückwärtslinearvorhersagekoeffizientenvektors b(k) des Eingangssignals x(k) und der Summe seiner a-posteriori-Vorhersagefehlerquadrate B(k), und

(b) wenn ein Vorfilterableitungskoeffizientenvektor durch f(k) repräsentiert sei, Ermitteln des Vorfilterkoeffizientenvektors g(k) durch eine Rekursionsformel, die sich aus der folgenden ersten und zweiten Gleichung zusammensetzt:

$$g(k) = (1-\mu)\begin{bmatrix} 0 \\ f(k-1) \end{bmatrix} + \frac{a(k)^T e(k)}{F(k)} a(k)$$

$$\begin{bmatrix} f(k-1) \\ 0 \end{bmatrix} = g(k-1) - \frac{b(k-1)^T e(k-1)}{B(k-1)} b(k-1)$$

4. Verfahren nach Anspruch 1 oder 3, bei dem, wenn das letzte Element des Vorfilterkoeffizientenvektors g(k-1) als durch $g_p(k-1)$ repräsentiert angenommen wird, der Vorfilterkoeffizientenvektor g(k) unter Verwendung der folgenden Gleichung berechnet wird, die eine modifizierte Version der zweiten Gleichung ist:

$$\begin{bmatrix} f(k-1) \\ 0 \end{bmatrix} = g(k-1) - g_p(k-1)\, b(k-1)$$

5. Verfahren nach Anspruch 1 oder 3, bei dem, wenn zwei vorbestimmte nicht-negative Zahlen durch $\delta_F(k)$ und $\delta_B(k-1)$ repräsentiert seien, der Vorfilterkoeffizientenvektor g(k) durch die folgenden Gleichungen berechnet wird, die modifizierte Versionen der ersten und der zweiten Gleichung sind:

$$g(k) = (1-\mu)\begin{bmatrix} 0 \\ f(k-1) \end{bmatrix} + \frac{a(k)^T e(k)}{F(k)+\delta_F(k)} a(k)$$

$$\begin{bmatrix} f(k-1) \\ 0 \end{bmatrix} = g(k-1) - \frac{b(k-1)^T e(k-1)}{B(k-1)+\delta_B(k-1)} b(k-1)$$

6. Verfahren nach Anspruch 2 oder 3, bei dem Schrift (v-5) das Berechnen des Korrelationsvektors $r_{p-1}(k)$ des Eingangssignals x(k) durch

$$r_{p-1}(k) = r_{p-1}(k-1)-x(k-L)x_{p-1}(k-L) + x(k)x_{p-1}(k)$$

umfaßt, wobei $x_{p-1}(k)$ den Vektor des Eingangssignals x(k) repräsentiert und ausgedrückt ist als:

$$x_{p-1}(k)=[x(k-1), x(k-2), ..., x(k-p+1)]^T.$$

7. Vorrichtung zum adaptiven Schätzen der Übertragungsfunktion eines unbekannten technischen Systems durch Verwendung eines Projektionsalgorithmus, wobei die Vorrichtung umfaßt:

Mittel zum Anlegen eines physikalischen Eingangssignals x(k) an das unbekannte System (12), wobei k eine diskrete Zeit repräsentiert;
Mittel zum Erfassen eines physikalischen Ausgangssignals y(k) aus dem unbekannten System (12);
Mittel (11) zum Schätzen der Übertragungsfunktion unter Verwendung des Eingangssignals x(k) und des Ausgangssignals y(k) und Erzeugen eines Ausgangssignals $\hat{y}$ (k) eines Schätzsystems (23) mit der geschätzten Übertragungsfunktion $\hat{h}$ (k), wobei das Schätzsystem als ein digitales Filter mit einer Anzahl L an Anzapfungen implementiert ist;
Mittel (24) zum Ermitteln, als ein Fehlersignal e(k)=y(k)-$\hat{y}$(k), der Differenz zwischen dem Ausgangssignal $\hat{y}$ (k) des Schätzsystems und dem Ausgangssignal y(k) des unbekannten Systems (12);
Mittel (21) zum Berechnen eines Korrigiervektors $\delta\hat{h}$ (k) für die geschätzte Übertragungsfunktion aus Eingangssignalvektoren x(k), x(k-1), ..., x(k-p + 1) und Fehlersignalen e(k), e(k-1), ..., e(k-p + 1), wobei p eine ganze Zahl größer oder gleich 2 ist und x(k) definiert ist als x(k) = [x(k), x(k-1), ..., x(k-L + 1)]^T ; und
Mittel (22) zum wiederholten Korrigieren, unter Verwendung des Korrigiervektors $\delta\hat{h}$ (k) und einer vorbestimmten Korrigierschringröße μ, der geschätzten Übertragungsfunktion $\hat{h}$ (k) zu jedem Zeitpunkt k durch die folgende Gleichung:

$$\hat{h}(k + 1)=\hat{h}(k)+\mu\delta\hat{h}(k)$$

so daß sich das Fehlersignal e(k) Null annähert;
dadurch gekennzeichnet, daß die Mittel (21) zum Berechnen des Korrigiervektors $\delta\hat{h}$ (k) umfassen:

- Mittel (31) zum Berechnen eines Vorfilterkoeffizientenvektors g(k) durch Lösen der folgenden simultanen linearen Gleichung mit p Unbekannten:

$$R_p(k)g(k)=e(k),$$

wobei e(k) den Vektor des Fehlersignals e(k) repräsentiert und definiert ist als

$$e(k)=[e(k), (1-\mu)e(k-1), ..., (1-\mu)^{p-1}e(k-p+1)]^T, \text{ und}$$

$R_p(k)$ eine Kovarianzmatrix des Eingangssignals x(k) ist und definiert ist als

$$R_p(k)=[x(k), x(k-1), ..., x(k-p+1)]^T [x(k), x(k-1), ..., x(k-p+1)];$$ und

- Mittel (32) zum Berechnen des Korrigiervektors $\delta\hat{h}(k)$ unter Verwendung des Vorfilterkoeffizientenvektors g(k) durch die folgende Gleichung:

$$\delta\hat{h}(k)=[x(k), x(k-1), ..., x(k-p+1)]\, g(k),$$

wobei x(k) den Vektor des Eingangssignals x(k) repräsentiert;
wobei die Mittel (31) zum Berechnen des Vorfilterkoeffizientenvektors g(k) umfaßt:

- Mittel (41) zum Berechnen eines Vorwärtslinearvorhersagekoeffizientenvektors a(k) des Eingangssignals x(k), der Summe seiner a-posteriori-Vorhersagefehlerquadrate F(k), eines Rückwärtslinearvorhersagekoeffizientenvektors b(k) des Eingangssignals x(k) und der Summe seiner a-posteriori-Vorhersagefehlerquadrate B(k), und
- Mittel (42, 43, 44) zum Ermitteln des Vorfilterkoeffizientenvektors g(k) durch eine Rekursionsformel, die sich aus der folgenden ersten und zweiten Gleichung zusammensetzt:

$$g(k) = (1-\mu)\begin{bmatrix} 0 \\ f(k-1) \end{bmatrix} + \frac{a(k)^T e(k)}{F(k)} a(k)$$

$$\begin{bmatrix} f(k-1) \\ 0 \end{bmatrix} = g(k-1) - \frac{b(k-1)^T e(k-1)}{B(k-1)} b(k-1)$$

wobei f(k) einen Vorfilterableitungskoeffizientenvektor repräsentiert.

8. Vorrichtung zum adaptiven Schätzen der Übertragungsfunktion eines unbekannten technischen Systems durch Verwendung eines Projektionsalgorithmus, wobei die Vorrichtung umfaßt:

Mittel zum Anlegen eines physikalischen Eingangssignals x(k) an das unbekannte System (12), wobei k eine diskrete Zeit repräsentiert;
Mittel zum Erfassen eines physikalischen Ausgangssignals y(k) aus dem unbekannten System (12);
Mittel (11) zum Schätzen der Übertragungsfunktion unter Verwendung des Eingangssignals x(k) und des Ausgangssignals y(k) und Erzeugen eines Ausgangssignals $\hat{y}(k)$ eines Schätzsystems (23) mit der geschätzten Übertragungsfunktion $\hat{h}(k)$, wobei das Schätzsystem als ein digitales Filter mit einer Anzahl L an Anzapfungen implementiert ist;
Mittel (24) zum Ermitteln, als ein Fehlersignal $e(k) = y(k) - \hat{y}(k)$, der Differenz zwischen dem Ausgangssignal $\hat{y}(k)$ des Schätzsystems und dem Ausgangssignal y(k) des unbekannten Systems (12); und
Mittel (20) zum Berechnen der geschätzten Übertragungsfunktion zu jedem Zeitpunkt k so, daß sich das Fehlersignal e(k) Null annähert;
dadurch gekennzeichnet, daß Mittel (20) zum Berechnen der geschätzten Übertragungsfunktion umfassen:
Mittel (31) zum Berechnen eines Vorfilterkoeffizientenvektors g(k) durch Lösen der folgenden simultanen linearen Gleichung mit p Unbekannten:

$$R_p(k)g(k)=e(k),$$

wobei e(k) den Vektor des Fehlersignals e(k) repräsentiert und definiert ist als

$$e(k) = [e(k), (1-\mu)e(k-1), ..., (1-\mu)^{p-1}e(k-p+1)]^T, \text{ und}$$

$R_p(k)$ eine Kovarianzmatrix des Eingangssignals x(k) ist und definiert ist als

$$R_p(k) = [x(k), x(k-1), ..., x(k-p+1)]^T [x(k), x(k-1), ..., x(k-p+1)];$$

- Mittel (51) zum Glätten von Vorfilterkoeffizienten $g_i(k)$, die Elemente des Vorfilterkoeffizientenvektors g(k) sind, durch die folgenden Gleichungen:

$$s_i(k)=s_{i-1}(k-1) + \mu g_i(k) \qquad \text{für } 2 \le i \le p$$
$$=\mu g_1(k) \qquad \text{für } i=1$$

zum Ermitteln eines Glättungskoeffizienten $s_i(k)$; und

- Mittel (55) zum Ermitteln einer ungefähren geschätzten Übertragungsfunktion z(k + 1) als eine Näherung der geschätzten Übertragungsfunktion unter Verwendung des Glättungskoeffizienten $s_p(k)$ auf der Basis der folgenden Gleichung

$$z(k + 1)=z(k) + s_p(k)x(k-p + 1);$$

wobei das Schätzsystem (23) umfaßt:

- Mittel (54) zum Berechnen der Faltung $x(k)^T z(k)$ der ungefähren geschätzten Übertragungsfunktion z(k) mit einem Eingangssignalvektor x(k), der definiert ist als $x(k) = [x(k), x(k-1), ..., x(k-L + 1)]^T$;
- Mittel (53) zum Berechnen des inneren Produkts $s_{p-1}(k-1)^T r_{p-1}(k)$, wobei der Vektor des Glättungskoeffizienten $s_i(k)$ und ein Korrelationsvektor $r_{p-1}(k)$ des Eingangssignals auf

$$s_{p-1}(k-1)=[s_1(k-1), s_2(k-1), ..., s_{p-1}(k-1)]^T,$$

bzw.

$$r_{p-1}(k) = [x(k)^T x(k-1), x(k)^T x(k-2), ..., x(k)^T x(k-p + 1)]^T,$$

gesetzt werden; und

- Mittel (57) zum Ermitteln der Summe des Faltungsergebnisses $x(k)^T z(k)$ und des inneren Produkts als das geschätzte Ausgangssignal $\hat{y}(k)$.

9. Vorrichtung nach Anspruch 8, bei der die Mittel (31) zum Lösen der simultanen linearen Gleichung mit p Unbekannten umfaßt:

- Mittel (41) zum Berechnen eines Vorwärtslinearvorhersagekoeffizientenvektors a(k) des Eingangssignals x(k), der Summe seiner a-posteriori-Vorhersagefehlerquadrate F(k), eines Rückwärtslinearvorhersagekoeffizientenvektors b(k) des Eingangssignals x(k) und der Summe seiner a-posteriori-Vorhersagefehlerquaduadrate B(k), und
- Mittel (42, 43, 44) zum Ermitteln des Vorfilterkoeffizientenvektors g(k) durch eine Rekursionsformel, die sich aus der folgenden ersten und zweiten Gleichung zusammensetzt:

$$g(k) = (1-\mu)\begin{bmatrix} 0 \\ f(k-1) \end{bmatrix} + \frac{a(k)^T e(k)}{F(k)} a(k)$$

$$\begin{bmatrix} f(k-1) \\ 0 \end{bmatrix} = g(k-1) - \frac{b(k-1)^T e(k-1)}{B(k-1)} b(k-1)$$

wobei f(k) einen Vorfilterableitungskoeffizientenvektor repräsentiert.

10. Vorrichtung nach Anspruch 8 oder 9, bei der die Mittel zur Berechnung des inneren Produkts Korrelationsrechenmittel zum Berechnen des Korrelationsvektors $r_{p-1}(k)$ des Eingangssignals x(k) durch

$$r_{p-1}(k)=r_{p-1}(k-1)-x(k-L)x_{p-1}(k-L) + x(k)x_{p-1}(k)$$

umfassen, wobei $x_{p-1}(k)$ den Vektor des Eingangssignals x(k) repräsentiert und ausgedrückt ist als:

$$x_{p-1}(k)=[x(k-1), x(k-2), ..., x(k-p + 1)]^T.$$

11. Vorrichtung nach Anspruch 7 oder 9, bei der das Vorfilterableitungskoeffizientenvektorkorrigiermittel Mittel sind, die, wenn das letzte Element des Vorfilterkoeffizientenvektors g(k-1) als durch $g_p(k-1)$ repräsentiert angenommen

wird, den Vorfilterkoeffizientenvektor g(k) unter Verwendung der folgenden Gleichung berechnen, die eine modifizierte Version der zweiten Gleichung ist:

$$\begin{bmatrix} f(k-1) \\ 0 \end{bmatrix} = g(k-1) - g_p(k-1)\, b(k-1)$$

12. Vorrichtung nach Anspruch 7 oder 9, bei der, wenn zwei vorbestimmte nicht-negative Zahlen durch $\delta_F(k)$ und $\delta_B(k-1)$ repräsentiert seien, die Vorfilterkoeffizientenvektorkorrigiermittel und die Vorfilterableitungskoeffizientenvektorkorrigiermittel den Vorfilterkoeffizientenvektor g(k) durch die folgenden Gleichungen berechnen, die modifizierte Versionen der ersten und der zweiten Gleichung sind:

$$g(k) = (1-\mu) \begin{bmatrix} 0 \\ f(k-1) \end{bmatrix} + \frac{a(k)^T e(k)}{F(k)+\delta_F(k)} a(k)$$

$$\begin{bmatrix} f(k-1) \\ 0 \end{bmatrix} = g(k-1) - \frac{b(k-1)^T e(k-1)}{B(k-1)+\delta_B(k-1)} b(k-1)$$

13. Vorrichtung nach Anspruch 7 oder 10, ferner umfassend Subtrahiermittel zur Bildung, als das Fehlersignal e(k), der Differenz y(k) - $\hat{y}$ (k) zwischen dem Ausgangssignal y(k) des unbekannten Systems (12) und dem geschätzten Signal $\hat{y}$ (k), welches das Ausgangssignal des Schätzsystems ist.

Revendications

1. Procédé d'estimation adaptative de la fonction de transfert d'un système technique inconnu par utilisation d'un algorithme de projection, ledit procédé comprenant :

   (i) l'application d'un signal physique d'entrée x(k) audit système inconnu (12), k représentant un temps discret ;
   (ii) la détection d'un signal physique de sortie y(k) provenant dudit système inconnu (12) ;
   (iii) l'estimation de la fonction de transfert en utilisant ledit signal d'entrée x(k) et ledit signal de sortie y(k) et en générant un signal de sortie $\hat{y}$ (k) d'un système estimé ayant ladite fonction de transfert estimée $\hat{h}$ (k), ledit système estimé étant réalisé sous la forme d'un filtre numérique ayant un nombre de prises de L ;
   (iv) l'obtention, en tant que signal d'erreur e(k) = y(k) - $\hat{y}$(k) , de la différence entre le signal de sortie $\hat{y}$ (k) dudit système estimé et le signal de sortie y(k) dudit système inconnu (12) ;
   (v) le calcul d'un vecteur de correction $\delta\hat{h}$ (k) pour la fonction de transfert estimée à partir de vecteurs de signaux d'entrée x(k), x(k-1), ..., x(k-p+1) et de signaux d'erreur e(k), e(k-1), ..., e(k-p+1) , où p est un entier égal ou supérieur à 2 et x(k) est défini par $x(k) = [x(k), x(k-1), ..., x(k-L+1)]^T$; et
   (vi) l'utilisation dudit vecteur de correction $\delta\hat{h}$ (k) et d'une taille de pas de correction prédéterminée μ pour corriger de façon répétée ladite fonction de transfert estimée $\hat{h}$ (k) à chaque instant k conformément à l'équation suivante :

$$\hat{h}(k+1) = \hat{h}(k)+\mu\delta\hat{h}(k)$$

   afin que ledit signal d'erreur e(k) tende vers zéro ;
   caractérisé en ce que l'étape (y) comprend les étapes :

   (v-1) de calcul d'un vecteur de coefficients de pré-filtre g(k) en résolvant le système d'équations linéaires à p inconnues suivant :

$$R_p(k)g(k) = e(k),$$

où e(k) représente le vecteur dudit signal d'erreur e(k) et est défini par :

$$e(k) = [e(k), (1-\mu)e(k-1),..., (1-\mu)^{p-1}e(k-p+1)]^T, \text{ et}$$

$R_p(k)$ est une matrice de covariance dudit signal d'entrée x(k) définie par :

$$R_p(k) = [x(k), x(k-1),...,x(k-p+1)]^T[x(k), x(k-1),..., x(k-p+1)] ; \text{ et}$$

(v-2) d'utilisation dudit vecteur de coefficients de pré-filtre g(k) pour calculer ledit vecteur de correction $\delta\hat{h}(k)$ conformément à l'équation suivante :

$$\delta\hat{h}(k) = [x(k), x(k-1), ..., x(k-p+1)]g(k),$$

où x(k) représente le vecteur dudit signal d'entrée x(k) ; l'étape (v-1) comprenant les étapes :
(v-1-a) de calcul d'un vecteur de coefficients de prédiction linéaire vers l'avant a(k) dudit signal d'entrée x(k), de la somme de ses carrés de l'erreur de prédiction a posteriori F(k), d'un vecteur de coefficients de prédiction linéaire vers l'arrière b(k) dudit signal d'entrée x(k) et de la somme de ses carrés de l'erreur de prédiction a posteriori B(k), et
(v-1-b) posant qu'un vecteur de coefficients de dérivation de pré-filtre est représenté par f(k), d'obtention du vecteur de coefficients de pré-filtre g(k) par une formule récursive constituée des première et seconde équations suivantes :

$$\mathbf{g}(k) = (1-\mu)\begin{bmatrix} 0 \\ \mathbf{f}(k-1) \end{bmatrix} + \frac{\mathbf{a}(k)^T\mathbf{e}(k)}{F(k)}\,\mathbf{a}(k)$$

$$\begin{bmatrix} \mathbf{f}(k-1) \\ 0 \end{bmatrix} = \mathbf{g}(k-1) - \frac{\mathbf{b}(k-1)^T\mathbf{e}(k-1)}{B(k-1)}\,\mathbf{b}(k-1)$$

2. Procédé d'estimation adaptative de la fonction de transfert d'un système technique inconnu par utilisation d'un algorithme de projection, ledit procédé comprenant :

(i) l'application d'un signal physique d'entrée x(k) audit système inconnu (12), k représentant un temps discret ;
(ii) la détection d'un signal physique de sortie y(k) provenant dudit système inconnu (12) ;
(iii) l'estimation de la fonction de transfert en utilisant ledit signal d'entrée x(k) et ledit signal de sortie y(k) et en générant un signal de sortie $\hat{y}$ (k) d'un système estimé ayant ladite fonction de transfert estimée $\hat{h}$ (k), ledit système estimé étant réalisé sous la forme d'un filtre numérique ayant un nombre de prises de L ;
(iv) l'obtention, en tant que signal d'erreur e(k) = y(k)- $\hat{y}$(k) , de la différence entre le signal de sortie $\hat{y}$ (k) dudit système estimé et le signal de sortie y(k) dudit système inconnu (12) ;
(v) la répétition du calcul de la fonction de transfert à chaque instant k afin que ledit signal d'erreur e(k) tende vers zéro ;
caractérisé en ce que
l'étape (y) comprend les étapes :

(v-1) de calcul d'un vecteur de coefficients de pré-filtre g(k) en résolvant le système d'équations linéaires à p inconnues suivant :

$$R_p(k)g(k) = e(k),$$

où e(k) représente le vecteur dudit signal d'erreur e(k) et est défini par :

$$e(k) = [e(k), (1-\mu)e(k-1),..., (1-\mu)^{p-1}e(k-p+1)]^T, \text{ et}$$

$R_p(k)$ est une matrice de covariance dudit signal d'entrée x(k) définie par :

$$R_p(k) = [x(k), x(k-1),..., x(k-p+1)]^T [x(k), x(k-1),..., x(k-p+1)] ;$$

(v-2) de lissage des coefficients de pré-filtre $g_i(k)$ qui sont des éléments d'un vecteur de coefficients de pré-filtre g(k) conformément aux équations suivantes :

$$s_i(k) = s_{i-1}(k-1) + \mu g_i(k) \quad \text{pour } 2 \le i \le p$$
$$= \mu g_1(k) \quad \text{pour } i=1$$

afin d'obtenir un coefficient de lissage $s_i(k)$;
(v-3) d'obtention d'une fonction de transfert estimée approximative z(k+1) en tant qu'approximation de ladite fonction de transfert estimée en utilisant le coefficient de lissage $s_p(k)$ sur la base de l'équation suivante :

$$z(k+1) = z(k) + s_p(k)x(k-p+1) ;$$

(v-4) de calcul de la convolution $x(k)^T z(k)$ de ladite fonction de transfert estimée approximative z(k) avec un vecteur de signal d'entrée x(k) défini par $x(k) = [x(k), x(k-1),..., x(k-L+1)]^T$;
(v-5) de calcul du produit scalaire $s_{p-1}(k-1)^T r_{p-1}(k)$, en fixant le vecteur dudit coefficient de lissage $s_i(k)$ et un vecteur de corrélation $r_{p-1}(k)$ dudit signal d'entrée respectivement à :

$$s_{p-1}(k-1) = [s_1(k-1), s_2(k-1), ..., s_{p-1}(k-1)]^T,$$

et

$$r_{p-1}(k) = [x(k)^T x(k-1), x(k)^T x(k-2), ..., x(k)^T x(k-p+1)]^T ; \text{ et}$$

(v-6) d'obtention de la somme dudit résultat de convolution $x(k)^T z(k)$ et dudit produit scalaire en tant que ledit signal de sortie estimé $\hat{y}(k)$.

3. Procédé selon la revendication 2, dans lequel l'étape (v-1) comprend :

(a) le calcul d'un vecteur de coefficients de prédiction linéaire vers l'avant a(k) dudit signal d'entrée x(k), de la somme de ses carrés de l'erreur de prédiction a posteriori F(k), d'un vecteur de coefficients de prédiction linéaire vers l'arrière b(k) dudit signal d'entrée x(k) et de la somme de ses carrés de l'erreur de prédiction a posteriori B(k), et
(b) posant que le vecteur de coefficients de dérivation de pré-filtre est représenté par f(k), l'obtention dudit vecteur de coefficients de pré-filtre g(k) par une formule récursive constituée des première et seconde équations suivantes :

$$\mathbf{g}(k) = (1-\mu)\begin{bmatrix} 0 \\ \mathbf{f}(k-1) \end{bmatrix} + \frac{\mathbf{a}(k)^T \mathbf{e}(k)}{F(k)} \mathbf{a}(k)$$

$$\begin{bmatrix} \mathbf{f}(k-1) \\ 0 \end{bmatrix} = \mathbf{g}(k-1) - \frac{\mathbf{b}(k-1)^T \mathbf{e}(k-1)}{B(k-1)} \mathbf{b}(k-1)$$

4. Procédé selon la revendication 1 ou 3, dans lequel, posant que le dernier élément dudit vecteur de coefficients de pré-filtre g(k-1) est représenté par $g_p(k-1)$, ledit vecteur de coefficients de pré-filtre g(k) est calculé en utilisant l'équation suivante qui est une version modifiée de ladite seconde équation :

$$\begin{bmatrix} \mathbf{f}(k-1) \\ 0 \end{bmatrix} = \mathbf{g}(k-1) - g_p(k-1)\mathbf{b}(k-1)$$

5. Procédé selon la revendication 1 ou 3, dans lequel, posant deux nombres non négatifs prédéterminés comme étant représentés par $\delta_F(k)$ et $\delta_B(k-1)$, ledit vecteur de coefficients de pré-filtre g(k) est calculé par les équations suivantes qui sont des versions modifiées desdites première et seconde équations :

$$\mathbf{g}(k) = (1-\mu)\begin{bmatrix} 0 \\ \mathbf{f}(k-1) \end{bmatrix} + \frac{\mathbf{a}(k)^T\mathbf{e}(k)}{F(k)+\delta_F(k)}\,\mathbf{a}(k)$$

$$\begin{bmatrix} \mathbf{f}(k-1) \\ 0 \end{bmatrix} = \mathbf{g}(k-1) - \frac{\mathbf{b}(k-1)^T\mathbf{e}(k-1)}{B(k-1)+\delta_B(k-1)}\,\mathbf{b}(k-1)$$

6. Procédé selon la revendication 2 ou 3, dans lequel l'étape (v-5) comprend le calcul dudit vecteur de corrélation $r_{p-1}(k)$ dudit signal d'entrée x(k) conformément à :

$$r_{p-1}(k) = r_{p-1}(k-1) - x(k-L)x_{p-1}(k-L) + x(k)x_{p-1}(k)$$

où $x_{p-1}(k)$ représente le vecteur dudit signal d'entrée x(k) exprimé par :

$$x_{p-1}(k) = [x(k-1), x(k-2), \ldots, x(k-p+1)]^T.$$

7. Dispositif d'estimation adaptative de la fonction de transfert d'un système technique inconnu par utilisation d'un algorithme de projection, ledit dispositif comprenant :

un moyen pour appliquer un signal physique d'entrée x(k) audit système inconnu (12), k représentant un temps discret ;
un moyen pour détecter un signal physique de sortie y(k) provenant dudit système inconnu (12) ;
un moyen (11) pour estimer la fonction de transfert en utilisant ledit signal d'entrée x(k) et ledit signal de sortie y(k) et en générant un signal de sortie $\hat{y}$ (k) d'un système estimé (23) ayant ladite fonction de transfert estimée $\hat{h}$ (k), ledit système estimé étant réalisé sous la forme d'un filtre numérique ayant un nombre de prises de L ;
un moyen (24) pour obtenir, en tant que signal d'erreur $e(k) = y(k) - \hat{y}(k)$, la différence entre le signal de sortie $\hat{y}$ (k) dudit système estimé et le signal de sortie y(k) dudit système inconnu (12) ;
un moyen (21) pour calculer un vecteur de correction $\delta\hat{h}$ (k) pour la fonction de transfert estimée à partir de vecteurs de signaux d'entrée x(k), x(k-1), ..., x(k-p+1) et de signaux d'erreur e(k), e(k-1), ..., e(k- p+1) , où p est un entier égal ou supérieur à 2, et x(k) est défini par $x(k) = [x(k), x(k-1), \ldots, x(k)\ x(k-L+1)]^T$; et
un moyen (22) pour corriger de façon répétée, à chaque instant k, en utilisant ledit vecteur de correction $\delta\hat{h}$ (k) et une taille de pas de correction prédéterminée μ, ladite fonction de transfert estimée $\hat{h}$ (k) conformément à l'équation suivante :

$$\hat{h}(k+1) = \hat{h}(k) + \mu\delta\,\hat{h}(k)$$

afin que ledit signal d'erreur e(k) tende vers zéro ;
caractérisé en ce que ledit moyen (21) de calcul du vecteur de correction $\delta\hat{h}$ (k) comprend :

- un moyen (31) pour calculer un vecteur de coefficients de pré-filtre g(k) en résolvant le système d'équations linéaires à p inconnues suivant :

$$R_p(k)g(k) = e(k),$$

où e(k) représente le vecteur dudit signal d'erreur e(k) et est défini par :

$$e(k) = [e(k), (1-\mu)e(k-1),..., (1-\mu)^{p-1}e(k-p+1)]^T \text{ , et}$$

$R_p(k)$ est une matrice de covariance dudit signal d'entrée x(k) définie par :

$$R_p(k) = [x(k), x(k-1),..., x(k-p+1)]^T[x(k), x(k-1),..., x(k-p+1)] \text{ ; et}$$

- un moyen (32) pour calculer, en utilisant ledit vecteur de coefficients de pré-filtre g(k), ledit vecteur de correction $\delta\hat{h}(k)$ conformément à l'équation suivante :

$$\delta\hat{h}(k) = [x(k), x(k-1), ..., x(k-p+1)]g(k)$$

où x(k) représente le vecteur dudit signal d'entrée x(k) ; et

ledit moyen (31) de calcul dudit vecteur de coefficients de pré-filtre g(k) comprend

- un moyen (41) pour calculer un vecteur de coefficients de prédiction linéaire vers l'avant a(k) dudit signal d'entrée x(k), la somme de ses carrés de l'erreur de prédiction a posteriori F(k), un vecteur de coefficients de prédiction linéaire vers l'arrière b(k) dudit signal d'entrée x(k) et la somme de ses carrés de l'erreur de prédiction a posteriori B(k), et
- un moyen (42, 43, 44) pour obtenir ledit vecteur de coefficients de pré-filtre g(k) par une formule récursive constituée des première et seconde équations suivantes :

$$\mathbf{g}(k) = (1-\mu)\begin{bmatrix} 0 \\ \mathbf{f}(k-1) \end{bmatrix} + \frac{\mathbf{a}(k)^T\mathbf{e}(k)}{F(k)}\,\mathbf{a}(k)$$

$$\begin{bmatrix} \mathbf{f}(k-1) \\ 0 \end{bmatrix} = \mathbf{g}(k-1) - \frac{\mathbf{b}(k-1)^T\mathbf{e}(k-1)}{B(k-1)}\,\mathbf{b}(k-1)$$

où f(k) représente un vecteur de coefficients de dérivation de pré-filtre.

8. Dispositif d'estimation adaptative de la fonction de transfert d'un système technique inconnu par utilisation d'un algorithme de projection, ledit dispositif comprenant :

un moyen pour appliquer un signal physique d'entrée x(k) audit système inconnu (12), k représentant un temps discret ;
un moyen pour détecter un signal physique de sortie y(k) provenant dudit système inconnu (12) ;
un moyen (11) pour estimer la fonction de transfert en utilisant ledit signal d'entrée x(k) et ledit signal de sortie y(k) et en générant un signal de sortie $\hat{y}(k)$ d'un système estimé (23) ayant ladite fonction de transfert estimée $\hat{h}(k)$, ledit système estimé étant réalisé sous la forme d'un filtre numérique ayant un nombre de prises de L ;
un moyen (24) pour obtenir, en tant que signal d'erreur $e(k) = y(k) - \hat{y}(k)$, la différence entre le signal de sortie $\hat{y}(k)$ dudit système estimé et le signal de sortie y(k) dudit système inconnu (12) ; et
un moyen (20) pour calculer la fonction de transfert estimée à chaque instant k afin que ledit signal d'erreur e(k) tende vers zéro ;
caractérisé en ce que ledit moyen (20) de calcul de ladite fonction de transfert estimée comprend :

- un moyen (31) pour calculer un vecteur de coefficients de pré-filtre g(k) en résolvant le système d'équations linéaires à p inconnues suivant :

$$R_p(k)g(k) = e(k),$$

où e(k) représente le vecteur dudit signal d'erreur e(k) et est défini par :

$$e(k) = [e(k), (1-\mu)e(k-1),..., (1-\mu)^{p-1}e(k-p+1)]^T \text{ , et}$$

$R_p(k)$ est une matrice de covariance et ledit signal d'entrée x(k) est défini par :

$$R_p(k) = [x(k), x(k-1),..., x(k-p+1)]^T[x(k), x(k-1),..., x(k-p+1)] \text{ ; et}$$

- un moyen (51) pour lisser les coefficients de pré-filtre $g_i(k)$, qui sont des éléments dudit vecteur de coefficients de pré-filtre g(k), conformément aux équations suivantes :

$$s_i(k) = s_{i-1}(k-1)+\mu g_i(k) \quad \text{pour } 2 \le i \le p$$
$$= \mu g_1(k) \quad \text{pour } i=1$$

afin d'obtenir un coefficient de lissage $s_i(k)$; et

- un moyen (55) pour obtenir une fonction de transfert estimée approximative z(k+1) en tant qu'approximation de ladite fonction de transfert estimée en utilisant le coefficient de lissage $s_p(k)$ sur la base de l'équation suivante :

$$z(k+1) = z(k) + s_p(k)x(k-p+1) \text{ ; et}$$

ledit système estimé (23) comprend :

- un moyen (54) pour calculer la convolution $x(k)^T z(k)$ de ladite fonction de transfert estimée approximative z(k) avec un vecteur de signal d'entrée x(k) defini par $x(k) = [x(k), x(k-1), ..., x(k-L+1)]^T$
- un moyen (53) pour calculer le produit scalaire $s_{p-1}(k-1)^T r_{p-1}(k)$, en fixant respectivement le vecteur dudit coefficient de lissage $s_i(k)$ et un vecteur de corrélation $r_{p-1}(k)$ dudit signal d'entrée, à :

$$s_{p-1}(k-1) = [s_1(k-1), s_2(k-1), ..., s_{p-1}(k-1)]^T,$$

et

$$r_{p-1}(k) = [x(k)^T x(k-1), x(k)^T x(k-2), ..., x(k)^T x(k-p+1)]^T \text{ ; et}$$

- un moyen (57) pour obtenir la somme dudit résultat de convolution $x(k)^T z(k)$ et dudit produit scalaire en tant que signal de sortie estimé $\hat{y}(k)$.

9. Dispositif selon la revendication 8, dans lequel ledit moyen (31) pour résoudre ledit système d'équations linéaires à p inconnues comprend :

un moyen (41) pour calculer un vecteur de coefficients de prédiction linéaire vers l'avant a(k) dudit signal d'entrée x(k), la somme de ses carrés de l'erreur de prédiction a posteriori F(k), un vecteur de coefficients de prédiction linéaire vers l'arrière b(k) dudit signal d'entrée x(k) et la somme de ses carrés de l'erreur de prédiction a posteriori B(k), et
un moyen (42, 43, 44) pour obtenir ledit vecteur de coefficients de pré-filtre g(k) par une formule récursive constituée des première et seconde équations suivantes :

$$\mathbf{g}(k) = (1-\mu)\begin{bmatrix} 0 \\ \mathbf{f}(k-1) \end{bmatrix} + \frac{\mathbf{a}(k)^T \mathbf{e}(k)}{F(k)} \mathbf{a}(k)$$

$$\begin{bmatrix} \mathbf{f}(k-1) \\ 0 \end{bmatrix} = \mathbf{g}(k-1) - \frac{\mathbf{b}(k-1)^T \mathbf{e}(k-1)}{B(k-1)} \mathbf{b}(k-1)$$

où f(k) représente un vecteur de coefficients de dérivation de pré-filtre.

10. Dispositif selon la revendication 8 ou 9, dans lequel ledit moyen de calcul de produit scalaire comporte un moyen de calcul de corrélation pour calculer ledit vecteur de corrélation $r_{p-1}(k)$ dudit signal d'entrée x(k) conformément à :

$$r_{p-1}(k) = r_{p-1}(k-1)-x(k-L)x_{p-1}(k-L)+x(k)x_{p-1}(k)$$

où $x_{p-1}(k)$ représente le vecteur dudit signal d'entrée x(k) exprimé par :

$$x_{p-1}(k) = [x(k-1), x(k-2), ..., x(k-p+1)]^T.$$

11. Dispositif selon la revendication 7 ou 9, dans lequel ledit moyen de correction de vecteur de coefficients de dérivation de pré-filtre est un moyen qui, posant que le dernier élément dudit vecteur de coefficients de pré-filtre g(k-1) est représenté par $g_p(k-1)$, calcule ledit vecteur de coefficients de pré-filtre g(k) conformément à l'équation suivante, qui est une version modifiée de ladite seconde équation :

$$\begin{bmatrix} \mathbf{f}(k-1) \\ 0 \end{bmatrix} = \mathbf{g}(k-1) - g_p(k-1)\mathbf{b}(k-1)$$

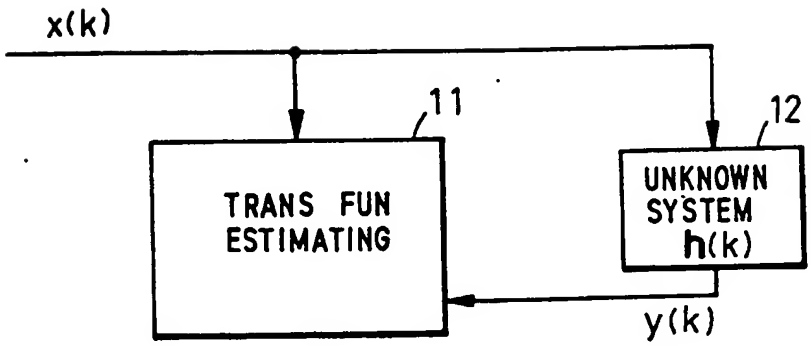
12. Dispositif selon la revendication 7 ou 9, dans lequel, posant que des nombres non négatifs prédéterminés sont respectivement représentés par $\delta_F(k)$ et $\delta_B(k-1)$, ledit moyen de correction de vecteur de coefficients de pré-filtre et ledit moyen de correction de vecteur de coefficients de dérivation de pré-filtre calculent ledit vecteur de coefficients de pré-filtre g(k) respectivement à partir des équations suivantes, qui sont des versions modifiées desdites première et seconde équations :

$$\mathbf{g}(k) = (1-\mu)\begin{bmatrix} 0 \\ \mathbf{f}(k-1) \end{bmatrix} + \frac{\mathbf{a}(k)^T\mathbf{e}(k)}{F(k)+\delta_F(k)}\,\mathbf{a}(k)$$

$$\begin{bmatrix} \mathbf{f}(k-1) \\ 0 \end{bmatrix} = \mathbf{g}(k-1) - \frac{\mathbf{b}(k-1)^T\mathbf{e}(k-1)}{B(k-1)+\delta_B(k-1)}\,\mathbf{b}(k-1)$$

13. Dispositif selon la revendication 7 ou 10, comprenant en outre un moyen de soustraction pour fournir, en tant que ledit signal d'erreur e(k), la différence y(k)- $\hat{y}$ (k) entre ledit signal de sortie y(k) dudit système inconnu (12) et ledit signal estimé $\hat{y}$ (k) qui est la sortie dudit système estimé.

FIG. 1 PRIOR ART



PRIOR ART

FIG. 2

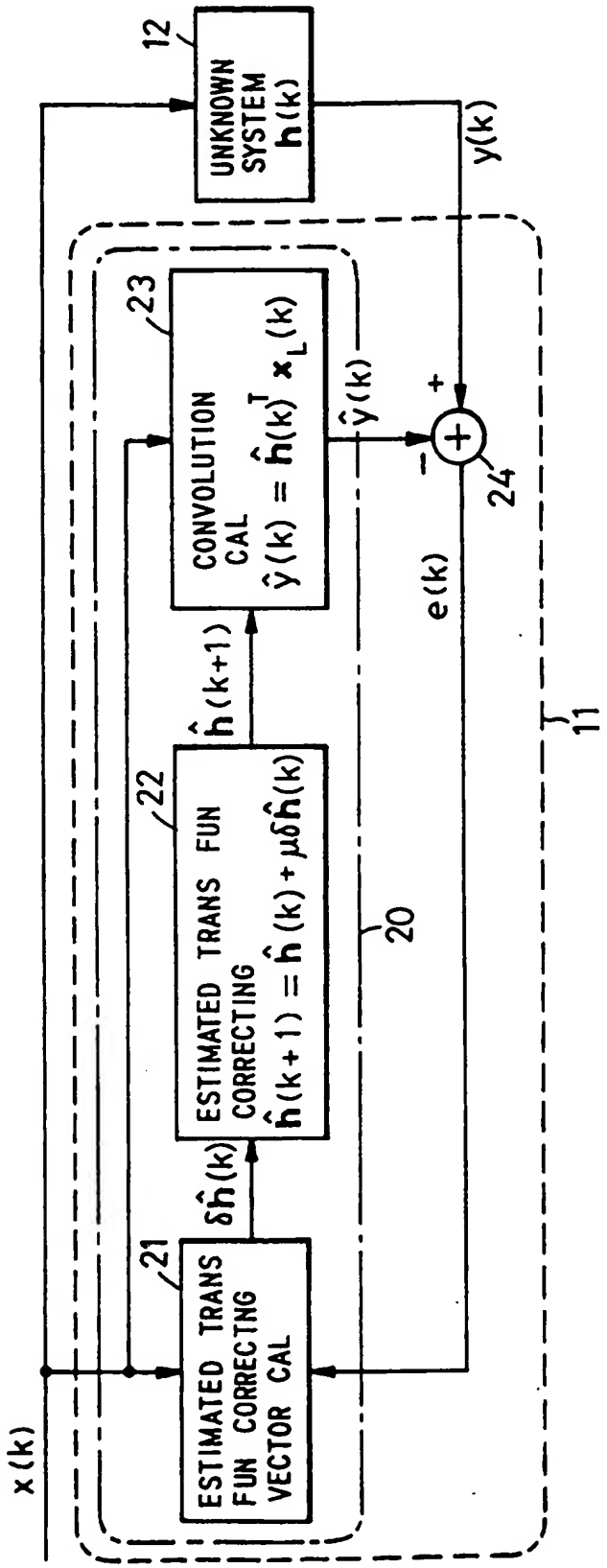
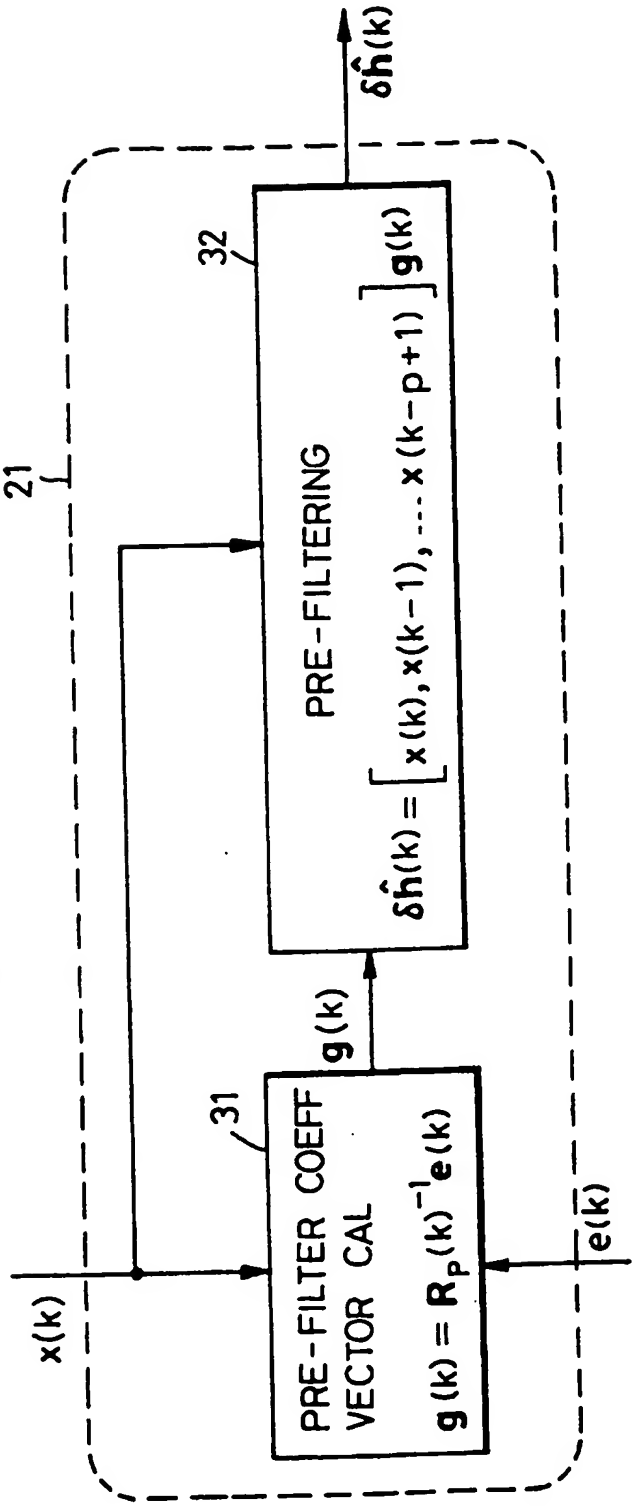


FIG. 3


21
32
PRE-FILTERING
$\delta\hat{\mathbf{h}}(k) = [\mathbf{x}(k), \mathbf{x}(k-1), \ldots \mathbf{x}(k-p+1)]\,\mathbf{g}(k)$
$\delta\hat{\mathbf{h}}(k)$
31
PRE-FILTER COEFF VECTOR CAL
$\mathbf{g}(k) = \mathbf{R}_p(k)^{-1}\mathbf{e}(k)$
$\mathbf{g}(k)$
$\mathbf{x}(k)$
$\mathbf{e}(k)$

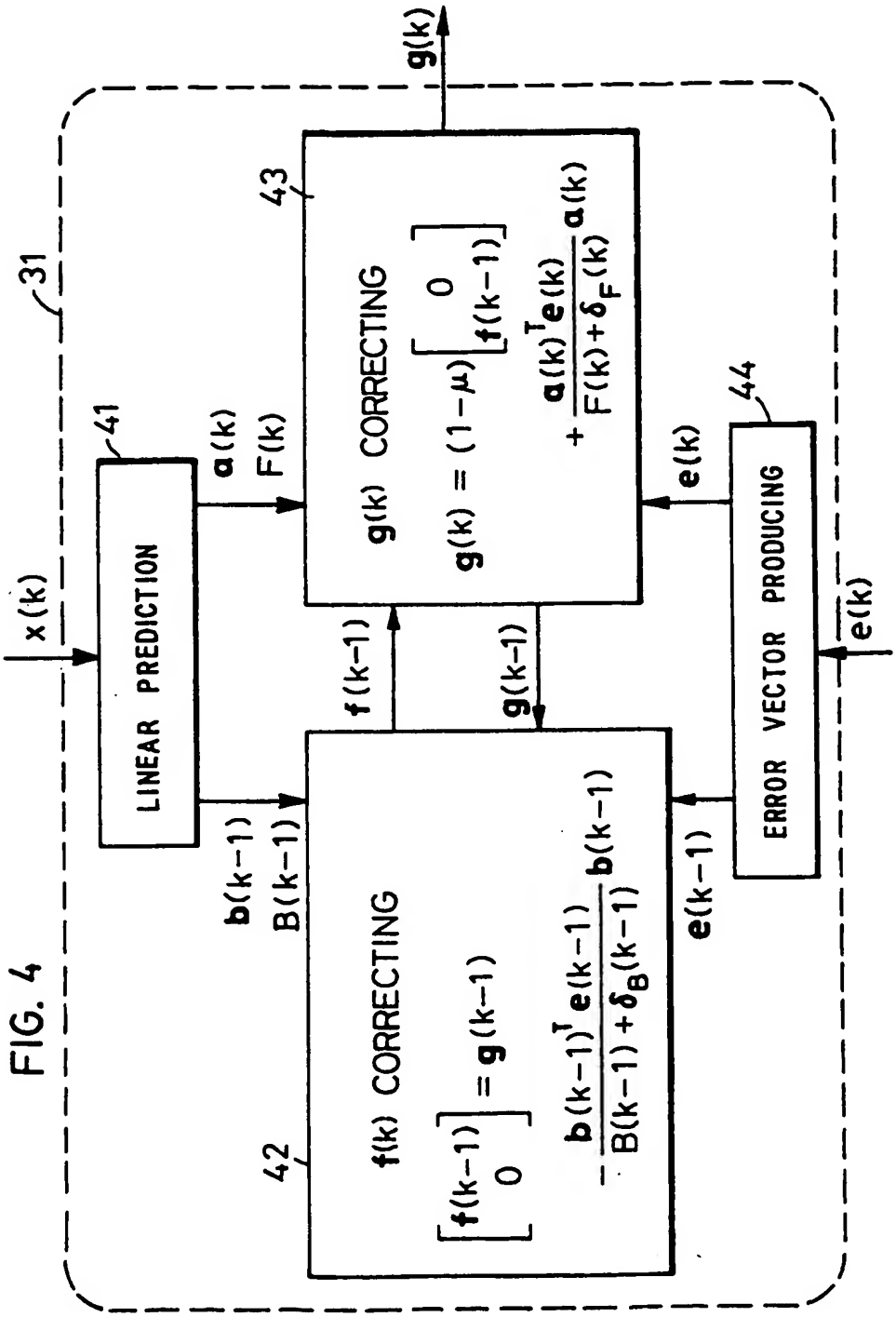

31
x(k)
41
LINEAR PREDICTION
a(k)
F(k)
b(k-1)
B(k-1)
42
f(k) CORRECTING
$\begin{bmatrix} f(k-1) \\ 0 \end{bmatrix} = g(k-1)$
$-\frac{b(k-1)^T e(k-1)}{B(k-1)+\delta_B(k-1)} b(k-1)$
f(k-1)
g(k-1)
43
g(k) CORRECTING
$g(k) = (1-\mu)\begin{bmatrix} 0 \\ f(k-1) \end{bmatrix}$
$+\frac{a(k)^T e(k)}{F(k)+\delta_F(k)} a(k)$
g(k)
e(k)
44
ERROR VECTOR PRODUCING
e(k-1)
e(k)

FIG. 4

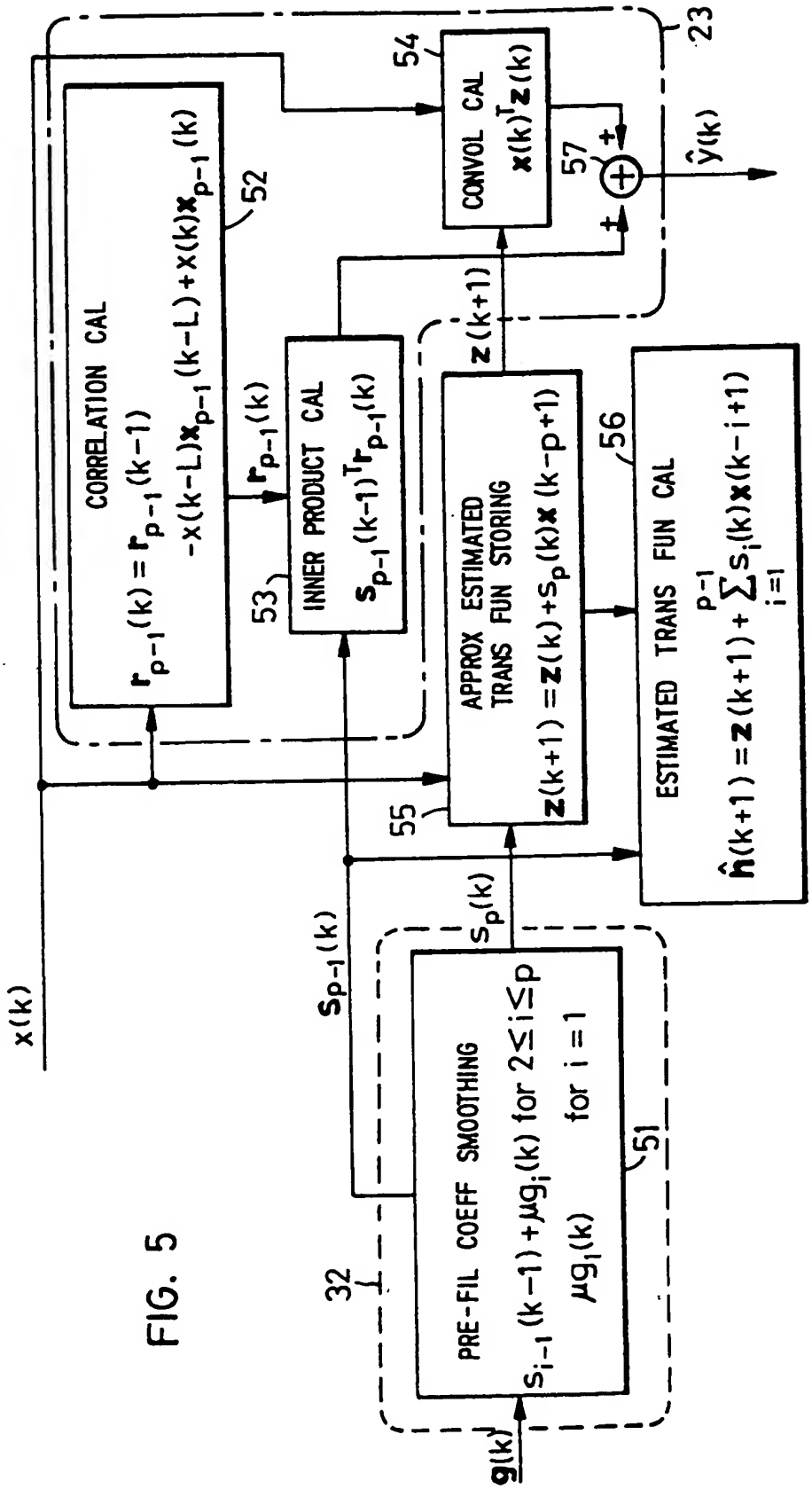

x(k)
g(k)
32
PRE-FIL COEFF SMOOTHING
$s_{i-1}(k-1)+\mu g_i(k)$ for $2\le i\le p$
$\mu g_1(k)$ for i = 1
51
$s_{p-1}(k)$
$s_p(k)$
CORRELATION CAL
$r_{p-1}(k)=r_{p-1}(k-1)$
$-x(k-L)x_{p-1}(k-L)+x(k)x_{p-1}(k)$
52
$r_{p-1}(k)$
53
INNER PRODUCT CAL
$s_{p-1}(k-1)^T r_{p-1}(k)$
55
APPROX ESTIMATED TRANS FUN STORING
$z(k+1)=z(k)+s_p(k)x(k-p+1)$
$z(k+1)$
56
ESTIMATED TRANS FUN CAL
$\hat{h}(k+1)=z(k+1)+\sum_{i=1}^{p-1} s_i(k)x(k-i+1)$
54
CONVOL CAL
$x(k)^T z(k)$
57
+
+
$\hat{y}(k)$
23

FIG. 5

FIG. 6

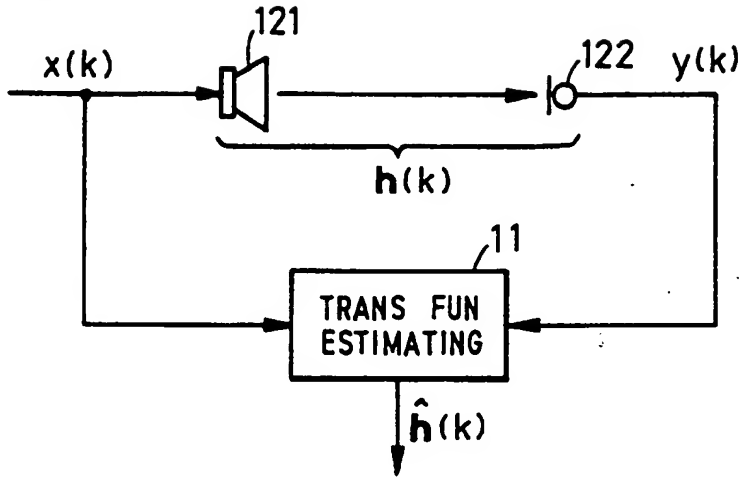

121
x(k)
122
y(k)
h(k)
11
TRANS FUN
ESTIMATING
ĥ(k)

FIG. 7

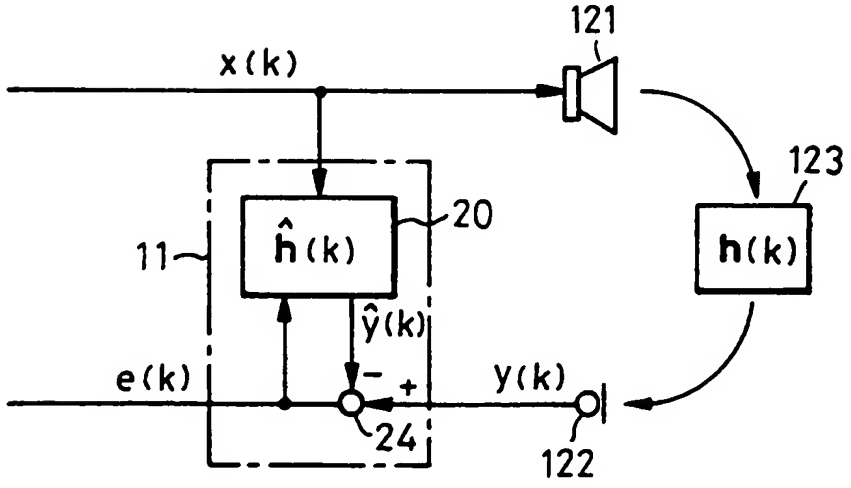

121
x(k)
123
20
11
ĥ(k)
h(k)
ŷ(k)
e(k)
y(k)
24
122

FIG. 8

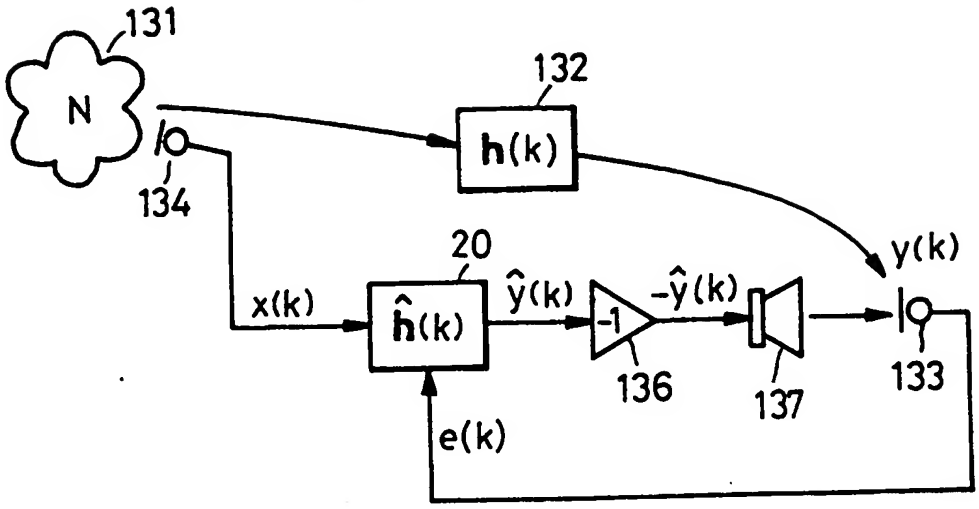

131
N
132
h(k)
134
20
y(k)
x(k)
ĥ(k)
ŷ(k)
-ŷ(k)
-1
136
137
133
e(k)

FIG. 10

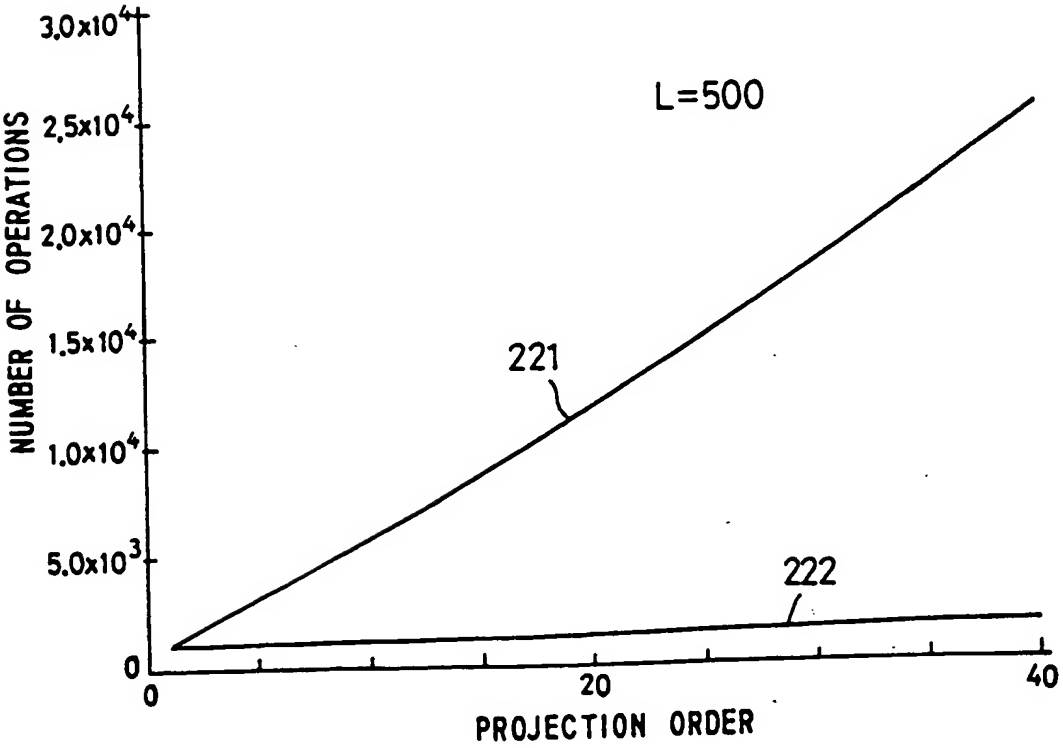
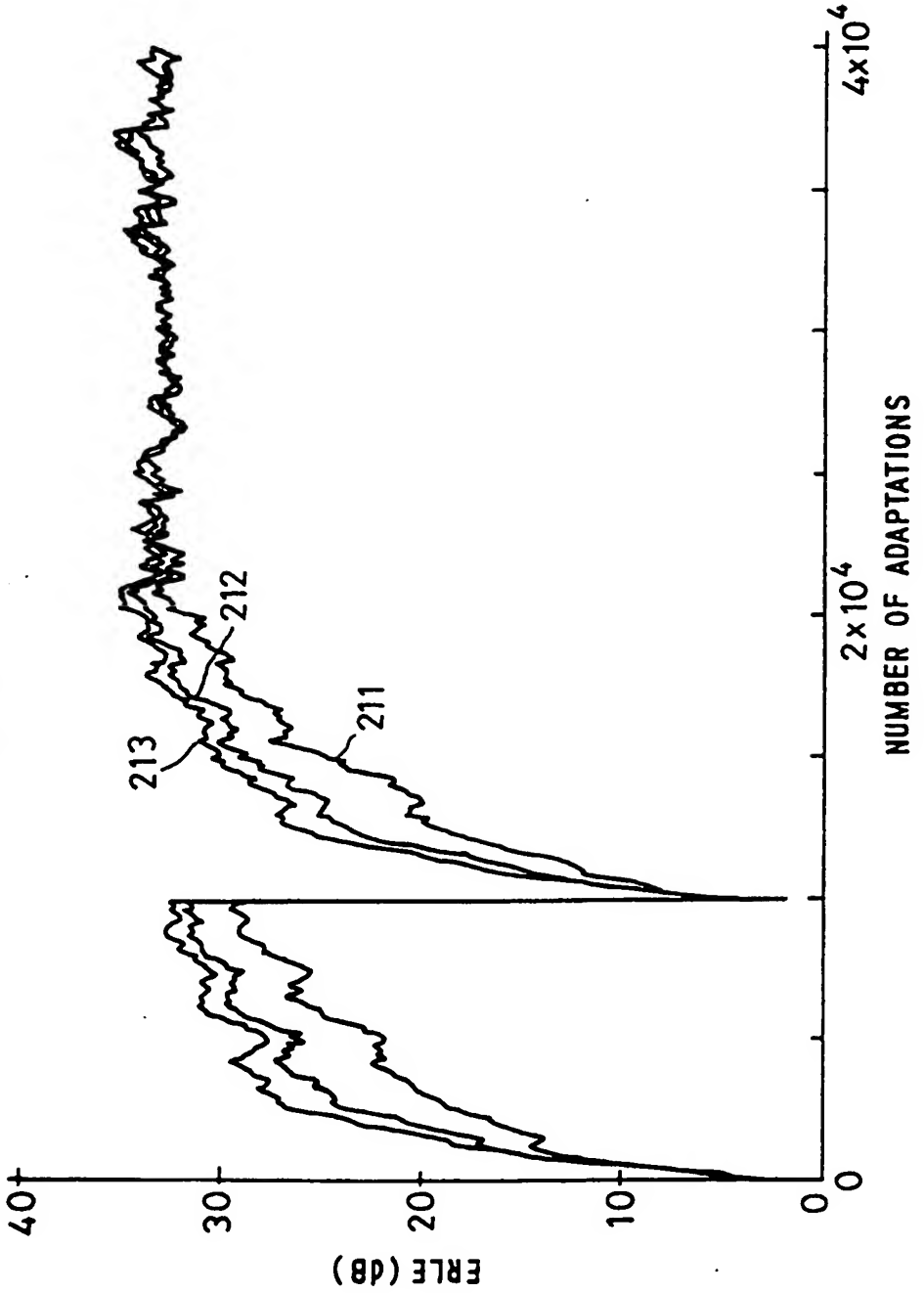

3.0x10^4
2.5x10^4
2.0x10^4
1.5x10^4
1.0x10^4
5.0x10^3
0
NUMBER OF OPERATIONS
L=500
221
222
0
20
40
PROJECTION ORDER

FIG. 9


ERLE (dB)
40
30
20
10
0
0
2x10^4
4x10^4
NUMBER OF ADAPTATIONS
213
212
211